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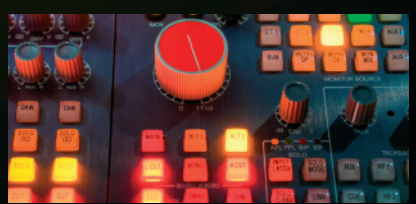

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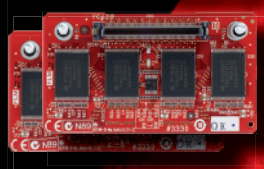

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The Art Of The Matter

It's a fact of life that some things are extremely difficult, if not impossible, to define. Take the concept of time, something that Einstein once referred to as 'a stubbornly persistent illusion'. Check its entry in any dictionary and you'll see that time is only ever defined using words that already imply an understanding of what time is, such as 'past', 'present', 'future' and 'order of occurrence of events'. Clearly, such definitions are meaningless, as they bring us no closer to understanding the true nature of time, even though we all have an instinctive grasp of how time applies to our own reality.

Equally complex is the subject of music. Although the mathematical relationships between notes are well understood and we have theory explaining rhythm and harmony, why we like some pieces of music and dislike others, or why certain musical forms evoke specific emotions, remains something of a mystery. We can compose music on an instinctive level, but if you try to devise a set of rules that allow a computer to make music the results are invariably disappointing, because we don't really know what those rules are.

A similar quandary exists when it comes to choosing sounds. I can go through a bank of synth patches or guitar tones and some will engage me on a musical level, while others leave me cold. Clearly established forms, such as the classical orchestra, sound 'right' because we've been exposed to them all our lives, but they had to evolve from simpler beginnings. I can see that if a sound mimics some familiar quality, such as a similarity to the human voice, we may find ourselves drawn to it, but beyond that, the appeal of harmonic makeup, envelopes and evolving timbres seems just as mysterious as any other aspect of music. Perhaps that's because music is art rather than science, but then definitions of art vary too. Some say that art can be anything that creates an emotion in the observer, in which case somebody sticking an ice pick in your arm would most certainly qualify. Maybe a better definition would include something along the lines of 'art is that which can be created without adherence to specific rules'?

When it comes to recording, there are aspects that have a scientific basis, such as not clipping your input converters, avoiding unwanted noise or arranging edits so as to avoid clicks, and we can explain the workings of recording and mixing techniques that have been shown to work in the past. But the reality for the recording musician is that they have to straddle the worlds of both science and art. This means that while it is possible to tell if a recording is technically OK, nobody is really qualified to tell you what you should record, or how it should sound when it's finished. After all, if your song makes everyone want to throw up or break furniture, then by at least one definition of art it has done its job!

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Sonar X1 launched

Cakewalk's DAW gets major update

Cakewalk came to the *SOS* offices recently to give us a sneak preview of the latest version of Sonar. Sonar X1, which should be available by the time you read this, incorporates a number of useful new features, many of which are designed to make working with the sequencer easier (or, as the press release says, for "streamlining your workflow").

Most visible of these is a new concept that Cakewalk call Skylight, which comprises a number of enhancements to Sonar's user interface. A new window, the MultiDock, can be set to show any of Sonar's views, including the Console, Arrange view, Piano Roll Editor, and even synth and effect interfaces. The window can be collapsed or expanded, and is designed to help minimise the need to switch between multiple windows when working. Cakewalk also say that it is ideal for people working with multiple monitors, as the MultiDock can be placed on a separate monitor so that frequently accessed views are always visible.

Then there's the Inspector pane, which displays data on clips, tracks and channels (whichever you have selected). With a channel highlighted, the Inspector will show not only the full channel strip (including effects and sends), but also that channel's output bus. The Inspector, like the MultiDock, can be expanded, collapsed, left floating or 'docked' to the top, bottom, left or right of the screen (the same is now true of all of Sonar's views and windows).

A Browser window has been incorporated, and this works in a similar way to the browsers that can be found in PreSonus' Studio One and Ableton's Live. It allows you to quickly find and audition a wide range of files, plug-ins and presets, including virtual instruments and effects, audio and MIDI files, track templates, song templates, and even entire projects. These can all be dragged and dropped from the browser into your current project — indeed, Cakewalk say that drag-and-drop functionality has been enhanced throughout Sonar, so you can now drag effects from the browser, from one channel to another, or even to individual audio clips. Another new addition is the Effects Chain, which allows you to group complex chains off effects, which can then be moved around your project (and dragged and dropped from the browser) in exactly the same way as a single plug-in.

Also new to Sonar X1 is the Smart Tool. A context-sensitive tool, this aims to reduce the amount of tool-changing needed by adapting its function depending on where the mouse is pointing (so you can cut, select and move items without switching tools). For those occasions where you do need to select a different tool, however, there's also a new Tool HUD (head-up display). This small floating window allows you to select any available tool (including the Smart Tool), and can be dragged anywhere on your screen, or be made to hide or appear with a single keystroke.

"The Pro Channel can be instantiated on any channel or bus, and comprises a compressor, an EQ and a valve saturation emulator."

Keyboard shortcuts have also been overhauled, and are now grouped around something called an Alpha Key, a key (normally a letter) around which all shortcuts for similar tasks are based. So, for example, pressing 'L' when you have a clip selected enables Looping for that clip. Pressing Shift+L sets the loop points to match your current Selection (a section of the time bar, or a group of clips, for example), while pressing Ctrl+L enables Groove Clip Looping for a selected audio clip.

As you can see, most of the enhancements are geared towards making Sonar easier and faster to work with, but there are some sonic treats thrown in with the latest version, most notably the Pro Channel. This can be instantiated on any channel or bus, and comprises a compressor, an EQ and a valve saturation emulator. There are two compressor types available, and by default, when the Pro Channel option is switched on for a channel, the compressor will be the PC76, an emulation of the classic FET levelling amplifier (no prizes for guessing which one!). When instantiated on a bus, however, the default compressor is one that Cakewalk call the PC4K S-Type, which is based on the famous bus compressor found in consoles from a well-known UK mixer manufacturer. The compressor can be switched between either type, however, whether inserted over a bus or a channel.

The Pro Channel also features Gloss EQ, a six-band parametric EQ that can work in

three different modes: Pure, Vintage and Modern. The control layout is similar to what you'd see on a mixer or channel strip, but EQ bands can also be adjusted using the graph that sits above the controls. The bands are configured as follows: there are two shelf/bell bands, four bell/notch bands, and two fully parametric filters (one high-pass and one low-pass). In addition, a button labelled Gloss provides "one-button shine", a slight 'smile' EQ curve designed to make things sound more exciting and up-front.

The saturation option can be switched on or off on a per-channel basis, and features two drive algorithms. The accompanying Drive control, meanwhile, ranges from "simple drive to full saturation".

A routing section in the Pro Channel allows you to place the valve saturation, compressor and EQ in any order. The entire Pro Channel can also be switched between pre- and post-insert operation.

Sonar X1 is available in three different versions, at three different price points. The 'full-fat' Sonar X1 Producer comes with all the bells and whistles, including the aforementioned Pro Channel, Session Drummer 3, Dimension Pro, Mastering Effects suite and True Pianos. Sonar X1 Studio, meanwhile, dispenses with the Pro Channel and some of the soft synths mentioned above, but is otherwise exactly the same as the Producer edition, and so should suit people who already have a collection of plug-ins and synths and therefore don't need some of the extras supplied with Producer. Sonar X1 Essential is the leanest version, and is compatible only with 32-bit versions of Windows (both the Producer and Studio versions will work on either 32- or 64-bit operating systems). It has a maximum track count of 64, and includes some of Sonar's basic content (including the Essential Instruments collection), as well as features like the Matrix view and Step Sequencer. For details about pricing, call Roland UK, or see the Sonar X1 web site.

Roland UK +44 (0)1792 702701
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PreSonus bargain starter pack

Getting your first recording rig together can be a bewildering, even daunting, experience, but PreSonus (www.presonus.com) have just announced a low-cost bundle of products that should make the experience much easier. It could even appeal to those who already have a home studio, but want to put together a mobile rig.

The PreSonus 1Box includes everything you need to get started, apart from a computer. Interfacing capabilities are provided by the two-in, two-out PreSonus AudioBox, which features two mic/DI inputs and two line outputs, plus a headphone output that mirrors the line outputs. The included PreSonus M7 condenser mic allows you to record acoustic sound sources, while a pair of HD7 headphones lets you monitor the results. A copy of PreSonus' Studio One Artist recording software is also bundled. Unlike many other entry-level DAWs, this offers unlimited tracks and plug-ins (computer horsepower permitting, of course).

To help the uninitiated get to grips with things, a copy of *Computer Recording Made Easy* is thrown into the bargain, while the whole package ships in an embossed metal case. But perhaps the 1Box's biggest appeal will be its price: just £199 including VAT. The PreSonus 1Box bundle is expected to be available later this month (just in time to make it onto your Christmas list!).

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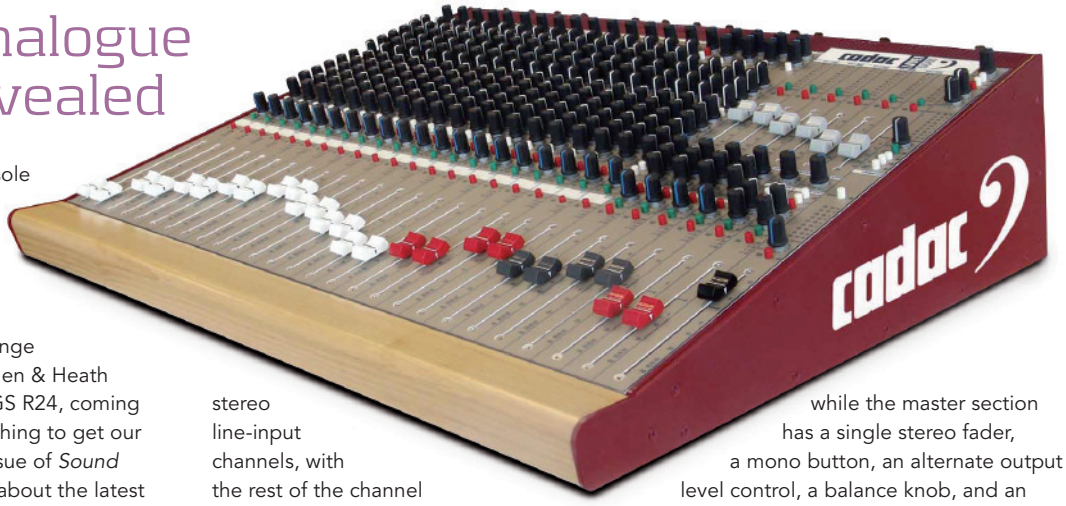
Cadac's compact console

Live 1 analogue mixer revealed



The analogue console seems to be undergoing something of a renaissance at the moment. Last month, we reported on a new range of mixers from Midas; Allen & Heath have a new model, the GS R24, coming out soon (which we're aching to get our hands on!); and in this issue of *Sound On Sound* you can read about the latest offerings from Audient (on page 20) and new company Neve Manufacturing Australia (page 86).

One name that might be less familiar to the average studio spod than those above is that of Luton-based manufacturers Cadac, and this month they, too, have announced a high-quality analogue mixing desk. Cadac's digital and analogue desks are well known in the live-sound and theatre worlds, and while their latest offering is designed primarily for live use (the clue is in the name: Live 1), Cadac say that this newcomer's sonic performance also makes it "the ideal analogue front end for DAWs".

Unlike Cadac's other mixers, which tend to be large to the point of being intimidating, the Live 1 is a compact affair, measuring just 570mm deep and either 480mm, 660mm or 840mm wide, depending on whether you opt for the 16-, 24- or 32-channel frame size (the smallest of which can be mounted in a rack). All mixers in the Live 1 range feature four stereo line-input channels, with the rest of the channel count being made up by mono mic/line strips. All mono channels feature a four-band EQ (comprising two swept mid-range bands and a high and low shelf), while the stereo channels have four fixed EQ bands. The mic preamps on the mono channels apparently employ "the latest engineering and component technologies", and are said by Cadac to be on a par with those of their large-format consoles.

A generous six auxes are available, with sends being switchable in pairs between pre- and post-fader operation, and aux send masters being controlled by 60mm faders above the master section. Six aux returns live just below the aux send faders, and four of these have pots for sending their signal to auxes 5 and 6 (for adding reverb to monitors or foldback, for example). There are a further two stereo returns, which, like the stereo and mono input channels, each have a 100mm fader. The four mono buses can be routed to the main stereo output and/or auxes 1 to 3, while the master section has a single stereo fader, a mono button, an alternate output level control, a balance knob, and an AFL Mode button, which makes all of the console's PFL buttons act post-fader.

Cadac are known for the rugged build quality of their desks, and the Live 1 is no exception. Although the top panel is constructed from a single metal sheet, internally the mixer is fully modular, which should make repairs significantly easier. All pots are nutted to the top panel (so you won't lose a pan pot inside the chassis if you accidentally lean on the desk!), and all of the Live 1's faders are Alps models. The technical spec also looks impressive: the common-mode rejection ratio given is -70dB, crosstalk figures are between -80dB and -90dB, dynamic range is quoted as a luxurious 119dB, and frequency response is apparently completely flat (±0dB!) between 5Hz and 65kHz.

Available from this month, pricing for the Cadac Live 1 starts at an estimated £4111 (including VAT) for the 16-channel version.

Cadac Holdings +44 (0)1582 404202
www.cadac-sound.com

Arturia Origin Keyboard

French synth makers Arturia (www.arturia.com) have unveiled the Origin Keyboard, a new instrument based on the Origin Rack but, as you may have deduced from its name, with a keyboard attached.

The synth part of the Origin Keyboard is pretty much identical to the Rack model. It sits on a hinged panel above a five-octave keyboard, and can be folded down when not in use. The keyboard features octave up and down buttons, mod and pitch wheels, and a freely assignable ribbon controller.

The Origin Keyboard is now shipping, and it carries a retail price of £2499. We'll be putting it through its paces in an *SOS* review soon!

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RME UFX announced

Thirty I/O via Firewire or USB!

RME have announced a new high-end multi-channel audio interface, called the UFX. A 1U-high rackmount device, the UFX provides both USB and Firewire 400 connectivity, so will be compatible with a wide variety of computers and laptops (including all Mac models). It offers a generous 30 inputs and 30 outputs in total, in a variety of digital and analogue formats. On the way in, there are eight line-only inputs, four mic/line inputs (these appear on the front panel for easy access) and two eight-channel ADAT inputs, plus a stereo AES3 digital input. Going the other way, there are eight line outs, two stereo headphone outputs (also located on the front panel), two eight-channel ADAT outs and a stereo AES3 output. One of the ADAT inputs and one ADAT output can optionally be used as optical S/PDIF I/O.

Usefully, two independent pairs of MIDI I/O are provided, one located on the front panel and another on the rear. A pair of BNC connectors provides word-clock I/O, while a front-panel USB port, labelled Memory, will eventually enable you to connect a USB drive for recording, even without a computer connected! This functionality hasn't been implemented yet, and details such as maximum track count weren't available at the time of writing, unfortunately, but they should be soon, and we'll cover them in detail in a forthcoming review.

Finally, a small nine-pin socket on the rear is for connecting an optional remote monitor controller, which we'll get onto later.

The four mic preamps are the same type as can be found in RME's Micstasy, and these digitally controlled affairs can supply up to 65dB of gain, the exact amount being governed either by the front-panel buttons and rotaries, or RME's TotalMix software. The front-panel XLR/jack 'combi' sockets can also accept either line or high-impedance instrument-level signals, and the nominal operating level for these can, again, be set either from the front panel or via TotalMix. Indeed, the same is true of all the UFX's analogue inputs and outputs: 0dBFS can be set so that it equates to an analogue level of +2dBV, +13dBu or +19dBu, on a per-channel basis, to ensure optimum level matching with your existing gear.

The A-D and D-A converters within are apparently completely new designs, and are capable of an impressive 115dB and 118dB dynamic range, respectively (both figures are A-weighted).

A large knob on the front functions as a monitor controller, and can be used to adjust the level of either the speaker outputs (which can be chosen from the front panel or within TotalMix), or, in conjunction with the front-panel buttons and menu system, the headphone outs. There's also a talkback function, where the talkback input can be chosen in TotalMix or from the front panel. On the right-hand side of the fascia resides a high-resolution display, which aids setup (most of the UFX's configuration options can be set here), and also provides accurate level metering.

The TotalMix software that comes with all of RME's interfaces has been significantly revised for the UFX, and is now called TotalMix FX. In addition to the routing and mixing functionality that all versions of TotalMix provide, TotalMix FX has a number of effects and signal processors built in, which run on DSPs inside the UFX rack. Every input and output channel benefits from a three-band parametric EQ, a variable high-pass filter, a compressor, an expander, a polarity-invert button, a trim control, an M/S tool, and an Auto-Level function. There are also built-in reverb and echo effects, which can be accesed by all channels via a send.

The control-room section has been updated, and now features a cue output for quick auditioning of any TotalMix outputs. There are four mute, solo and fader groups, and all channels can be individually switched between mono and stereo operation. User interface enhancements include a two-row mode, which reduces the mixer height to two rows for smaller displays; a slim-channel mode, which horizontally collapses selected channels to save space, or to allow TotalMix FX to function as a meter bridge; and the ability to expand any channel to show EQ and dynamics-processing settings.

The aforementioned remote control is a compact affair, providing monitor level control (including dim switching), as well as level-setting storage and routing settings recall. No price was available for the remote control at the time of writing, unfortunately, but we do know that the main UFX unit will retail for £1799 including VAT. For more information, check out the RME web site.

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The ISA828 eight channel microphone pre-amplifier provides eight original ISA transformer-based preamps in a single robust 2U chassis, offering the Focusrite signature sound at its lowest cost per channel to date. The same eight channel A-D converter option is also available.

ISA One Analogue and Digital - Typically £399.99 and £599.99

ISA One is ideal for both engineers and demanding performers alike, featuring the classic ISA transformer-based pre amp, a simultaneous DI, an output for routing to an amp, an independent XLR output on the rear and, on the Digital option, routing to the digital output.

www.focusrite.com/try

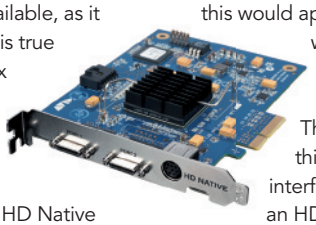
*Terms and conditions apply. Speak with your nearest Focusrite pro-dealer for details.

Photograph taken at Phoenix Sound Limited, Pinewood Studios

Pro Tools HD goes native!

Avid's flagship DAW now runs without TDM

Avid have announced a new version of their flagship DAW, Pro Tools HD, which runs on the host computer's CPU. Pro Tools HD Native has an almost identical feature set to the existing TDM-powered Pro Tools HD, including full plug-in delay compensation, support for Avid's Icon, D-Command and C24 control surfaces, surround-sound mixing and monitoring, and, just like Pro Tools HD, support for up to 192 audio tracks, 160 auxes and 128 instrument tracks. In fact, one of the only limitations of HD Native compared with the TDM-powered version is physical I/O count, though HD Native is still no slouch in that department (of which more later). The other most notable restriction of HD Native is that HEAT, the mixer add-on recently released by Avid, is not available, as it requires dedicated DSP processing to run. The same is true of all TDM-only plug-ins, such as Cranesong's Phoenix tape emulator.



All audio signal processing in Pro Tools HD Native is done on the CPU, and not by Avid's Core or Accel TDM cards. However, the new version does require Avid hardware to run. Specifically, a Pro Tools HD Native PCIe card must be installed in the host computer, and this, like the TDM Core card, allows you to connect Digilink-compatible audio interfaces, such as the new HD Omni box, or the older 192 IO and 96 IO interfaces.

The HD Native PCIe card features two Mini-Digilink ports, whereas the TDM Core and Accel cards feature just one full-size Digilink socket. This, combined with the fact that Pro Tools HD interfaces can be daisy-chained, means that four Avid interfaces can be connected at a time, for a total of 64 physical inputs and outputs (each HD interface is capable of providing 16 simultaneous ins and outs).

It should be noted that although audio processing in Pro Tools HD Native is performed by the CPU, the HD Native PCIe card does carry out such functions as calculating plug-in delay and facilitating direct input monitoring, in addition to providing interface connectivity.

Logically enough, the processing capabilities of Pro Tools HD Native are directly tied to the power available in the host Mac or PC, which, given the performance of the latest generation of processors, is potentially vast. However, there are occasions when the guaranteed amount of processing that dedicated DSP chips bring is advantageous, such as when opening projects on different systems. For this reason, Avid say that they will continue to sell and support DSP-based Pro Tools HD systems and hardware.

Pro Tools HD Native is available in a variety of differently priced bundles. The card and DAW software together costs £2500, and this would appear to be suitable for existing Pro Tools HD users who already own an Avid or Digidesign interface, or for people wishing to invest in one of the third-party options, such as those provided by Prism or Apogee. The cheapest 'complete' bundle costs £4300, and this includes the card, the software and an HD Omni interface. Next up is the 8x8x8 bundle, which comes with an HD IO instead of the Omni (the HD IO in this instance is configured with eight analogue inputs and outputs), and costs £5000. And finally, the 16x16 bundle, which costs £5750, ships with a fully loaded HD IO, capable of sending and receiving 16 analogue channels simultaneously.

Though hardly cheap, the new Pro Tools HD Native bundles do represent a significant saving over TDM-powered Pro Tools systems. For example, in order to connect four Avid interfaces to a TDM HD rig, it would be necessary to have at least one Core TDM card and one Accel card, which together are much more costly than the Native solution. The host-based nature of Pro Tools HD Native also means that, computer permitting, you get the same number of Voices (channels, auxes and so on) as would require a HD3 rig to achieve, which, again, would cost significantly more.

For more information on Pro Tools HD Native, including a chart comparing latency figures for Native and TDM systems, check out the Avid web site.

Avid UK +44 (0)1753 655999
www.avid.com

Analogue on a diet

Moog launch SlimPhatty synth

Hot on the heels of the launch of their new semi-modular synth, the Minimoog Voyager XL, Moog Music have sent us news of a rather more compact instrument: the SlimPhatty. The Moog SlimPhatty is, in essence, the sound-generating part of a Little Phatty, but with the keyboard omitted. It features two oscillators, a fourth-order (24dB/octave) Moog ladder filter, two ADSR envelopes and a four-waveform LFO. A 1x2 Mod Bus allows you to route the eight possible modulation sources to four different destinations, while four control voltage inputs (for volume, filter, pitch and gate) allow you to integrate external synths and modules.

Best of all, however, is the price: this bijou box is a bit of a bargain, at just £699 including VAT! The Moog Music SlimPhatty is expected to be available in the UK from the beginning of next year. For more information, check out the Moog Music web site, below.

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Charter Oak PEQ1

Stereo mastering equaliser

Charter Oak's foray into the world of outboard continues, with the announcement this month that their new equaliser, the PEQ1, will be shipping imminently. The device is a 2U stereo processor that its makers say "is designed to enhance your stereo mixes more than any other single component in your signal path". This, combined with the ganged filter controls (there can be no dual-mono operation, sadly), suggests that it was designed with mastering in mind.

The inputs and outputs are transformer-balanced, and both have a distinctly old-school impedance of 600Ω. Left and right gain trim pots can be found on the right-hand side of the front panel, while over at the far left are a bypass switch and a toggle for engaging a first-order high-pass filter with a turnover frequency of 40Hz.

Gain ranges on the sweepable bands appear to have been well thought out: towards the extreme ends of the frequency spectrum, where our ears are less sensitive, bands can be cut or boosted quite extremely, while the amount of gain available on the mid-range bands, where our hearing is more acute, has been reduced. Six bands are available, each with a switch to select between one of either two or three frequencies, and all bands have a fixed two-octave bandwidth.

The bottom band can be switched between 20Hz, 40Hz and 60Hz. The next one is switchable between 80Hz and 120Hz, and the next up from that between 400Hz, 630Hz and 800Hz. A toggle switch for choosing 1kHz, 2kHz or 3kHz resides below the next gain knob, while to the right of that you can choose between 5kHz and 8kHz. Finally, the top band can be set to 15kHz, 20kHz or a bat-bothering 50kHz. Gain ranges are ±18dB for the very top and bottom bands, ±9dB for the bands immediately adjacent to the top and bottom bands, and ±4.5dB for the two bands in the middle.

Charter Oak's UK distributors, ASAP Europe, expect the PEQ1 to be available from November, at a price of £2595 including VAT. We'll be putting it through its paces in *SOS* as soon as we can get our mitts on one!

ASAP Europe +44 (0)20 7231 9661
www.asapeurope.com
www.charteroakacoustics.com

SSR, the **School of Sound Recording**, have just announced that they will be opening a second campus, in celebration of their 30th anniversary. The new facility is based in Camden, North London, in an old piano factory on Gloucester Crescent. SSR London will open this coming January, and promises to be a state-of-the-art facility. The main studio will house a Neve Legend VR60, a Studer multitrack tape recorder and "a plethora of vintage microphones and outboard". Other facilities include a 5.1 surround-sound suite based around an Avid Icon controller, a "Soundcraft studio" (the mixer model isn't given), a Mac-based suite and a Foley recording studio.

SSR London will be providing a wide range of audio courses, including certified courses in Pro Tools and Logic. As a special treat, they'll be hosting a series of special events and free masterclasses in January. For more information, check out the SSR web site.

www.s-s-r.com

Rob Papen RP-Delay now available

Rob Papen (www.robpapen.com), the man behind many of the sounds from the Access Virus and Alesis Andromeda synths, has unveiled a new delay plug-in. RP-Delay is a rather more complex affair than your common-or-garden delay effect — indeed, many of its controls are more like the kind of thing you'd find on a synth!

Of course, it does simple mono and stereo delays, and there's a tape-delay emulation option, which even allows you to change the virtual tape length in real time without any clicking or popping. But more unusual is the inclusion of four envelopes, four LFOs and four modulation sequencers. There's also an interesting Reverse function, which not only reverses the delay, but can reverse the dry signal as well. What's more, the reversed sound can be triggered via keyboard (as can some of the modulation options), so it seems the plug-in can be made to work in quite an instrument-like way.

RP-Delay is available now on its own for £42, and it also comes bundled with Papen's RP-Verb plug-in. It is available in VST, RTAS and AU formats, and can be trialled for free, for 30 days. To find out more, check out the Rob Papen web site or www.timespace.com.

Last month, we brought you news of **Novation**'s new Ultranova synth. In that story, we said that the Ultranova can act as a MIDI interface, which is nearly true, but not quite. The Ultranova can send MIDI data from the keyboard and synth via USB, and it can receive and act on MIDI data received via USB, so there is MIDI USB functionality, but Novation have asked us to point out that this can't be classified as a USB MIDI interface, because this data can't be sent elsewhere — it can't transmit MIDI data received via USB to the five-pin DIN MIDI output, neither can it pass MIDI information received via the MIDI input to USB.

www.novation.com

Music course providers **Point Blank** have announced a new one-year programme, the Intensive Music Production & Sound Engineering Diploma. A full-time course, the diploma comprises six modules (which are taken two at a time) that cover a wide range of topics. The modules are: Introduction To Music Production, Music Composition, Sound Design, Sound Engineering, Art Of Mixing, and Music Industry Project (this last one being a kind of 'final project' that puts into practice what you have learnt over the rest of the course). Students each get the use of a Mac-based workstation during classes, as well as access to Point Blank's

Brainworx's new side-splitter

BX XL multi-band limiter released

Mid/Side processing experts Brainworx have released a new limiter plug-in, called BX XL. Like many of their products, including BX Digital and, more recently, the BX ShredSpread, the BX XL is made for processing stereo signals. It splits the left and right channels into Mid and Side for processing, then decodes the Mid/Side signals back into stereo. The BX XL also splits the Mid component of your programme material into two different frequency bands, allowing independent limiting of high and low frequencies in the Mid channel, as well as broadband limiting of the Side channel.

A Mono Maker control, of the same kind that can be found in many of Brainworx's other products, collapses everything below a given frequency to mono, to ensure a stable stereo image at low frequencies, as well as taking care of vinyl compatibility.

After the Mid/Side splitting and multi-band limiting, the decoded stereo signal passes through a fourth broadband limiter to further reduce dynamic range. Mid and Side signal levels can be adjusted post-processing, for manipulating stereo width, while copious Solo buttons allow you to listen to each part of the audio you're processing, such as the upper band of the Mid component, or the broadband Side signal, for example.

Available now, the Brainworx BX XL plug-in carries a normal price of €299, but if you buy it before November 25th, you'll get it for the significantly reduced price of €199. The plug-in can also be trialled for free, for 14 days, without an iLok dongle.

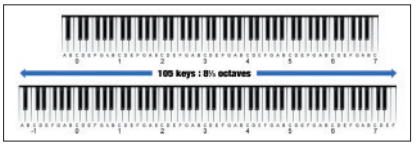
Brainworx +49 (0)2173 204 18 52
www.brainworx-music.de

studio facilities, seven days a week, for the duration of the course. Enrolment is taking place now, with the first two courses starting on January 10th and April 4th next year. The one-year Intensive Music Production & Sound Engineering Diploma costs £8990.
www.pointblanklondon.com

SOS Editor In Chief **Paul White** was recently presented with a Gold Badge Award by the British Academy of Songwriters, Composers and Authors (BASCA), at the organisation's 37th annual award ceremony. The Gold Badge Award is given to people who are deemed to have made a special contribution to the music industry in Britain, and was awarded to Paul for the numerous books and articles he's written on the subject of recording (including many, of course, for *Sound On Sound*). So, on behalf of everyone who's ever learnt a thing or two from Paul's writing, we'd like to say thanks, and congratulations!
www.basca.org.uk

Stephen Bull, owner of microphone manufacturers Newmann Retro, has announced a new venture, called SB Audio (named, obviously, after Stephen Bull). The new venture, Stephen tells us, is not only a music production company (they will apparently offer services such as remixing, mastering, post-production and tape restoration), but it will also be the brand under which all of Stephen's new products will be released. The current Newmann products (including the Newmann Retro and the NR47, both of which we have reviewed in *SOS*) will keep their original Newmann branding, to avoid confusion. SB Audio say that they will be launching their new web site next year, but in the meantime, some information on the services they offer can be found at the Newmann Retro web site, below.
www.newmannretro.com

Modartt, the French software developers who brought us the Pianoteq software instrument, have announced that their K1, C3 and M3 grand piano



emulations have had their ranges extended to a monstrous 105 keys — more than any real piano in the world! This feat is made possible by the fact that, instead of using hard-disk-consuming samples, Modartt generate their piano sounds in real time using a physical modelling process. The extended key range comes with the latest version of Pianoteq (v3.6.5), which is available now and free to all existing Pianoteq owners.
www.pianoteq.com

AT's first impressions

Around a year ago, Audio Technica (www.audio-technica.com) announced a scheme called First Impressions, which allowed customers to try out any mic in their 40 Series, Artist Series and Artist Elite product ranges, for two weeks, before committing to buy. Now the company have announced that a range of their headphones can also be auditioned under the same scheme: the popular ATH M50, ATH W1000X and ATH W5000 headphones can now be tried for free, for two weeks, allowing you to put them through their paces before deciding to buy a pair. More information about Audio Technica's First Impressions scheme can be found at www.eu.audio-technica.com/firstimpressions.

Audient ASP2802

Analogue Console & DAW Controller

Offering hands-on DAW control *and* a 32-input analogue console with bus compressor, Audient's latest mixer is very impressive indeed — and it's not as expensive as you might expect...

HUGH ROBJOHNS

Nobody would deny the creative power and supreme flexibility afforded by the modern Digital Audio Workstation (DAW), but controlling such systems through a keyboard and mouse is often far from ideal — particularly when it comes to more tactile operations, such as mixing and adjusting audio parameters.

This, of course, is where the hardware control surface comes into play, a wide variety of which have become available over recent years. These can range from neat, handheld and wireless units with a single assignable fader and basic transport controls, through to Euphonix's immensely configurable flagship EuCon controller, the MC Pro, and other high-end devices such as the Control 24 for Pro Tools and Smart AV's Smart Console. The middle ground is dominated by devices that mostly operate under the Mackie Control and HUI protocols, and although there are some frustrating variations in how some DAWs implement these protocols,

by and large they offer a reliable and familiar control interface.

The Best Of Both Worlds

Running in parallel with this common modern-day requirement for a hardware control surface of some kind remains a keen interest in external analogue summing or mixing hardware, along with the obvious practical requirement for some kind of front end (mic preamps) and back end (monitoring chain) hardware to get audio in and out of the DAW conveniently and effectively. Several manufacturers have developed equipment to address these points in various ways — either as isolated sub-systems or, in some cases, in the form of integrated console/controllers.

The likes of SSL and Allen & Heath have chosen the latter path, and another British company, Audient, have also adopted this approach. Audient's first foray into this complex and competitive field was the introduction of a mechanical variant of their ASP8024 console, which included panel space in the centre to install a standard Mackie HUI hardware controller (or similar), thereby allowing convenient control of a DAW from the console, albeit without any level of integration beyond the mechanical.

The next product on this evolutionary line was Audient's Zen console (reviewed in *SOS* December 2009) which provided a cut-down — but still very powerful — analogue console for recording, mixing and monitoring. This time, though, it featured fully integrated (albeit somewhat limited) DAW control and recall facilities: the fader movements of the analogue console could be used to control a DAW's fader movements and record level automation; and a DAW could also be used to record and play back fader movements to control an analogue mix on the console. The Zen also included dedicated transport buttons for DAW control.

The latest stage in Audient's progressive development of integrated DAW control is their new ASP2802 console, reviewed here (running firmware v1.01), which borrows its analogue technology from the Zen but raises the DAW control-system bar to a whole new level. It substantially extends and expands on the Zen's capabilities, while simultaneously coming in at a vastly reduced price.

The illuminated Cut and Solo buttons above each channel's fader can affect either the DAW or the analogue signal paths.

Audient ASP2802
£3760

PROS

- A very compact, but enormously versatile console.
- 32 inputs to mixdown.
- Superb quality analogue electronics with great I/O flexibility.
- All balanced analogue I/O on sensible connectors.
- No integrated computer interface — the user can select their own.
- Professional standard monitoring and talkback facilities.
- Routable bus compressor.
- Well thought-out and comprehensive DAW-control features.
- Attractive OLED DAW-control displays.

CONS

- No integrated computer interface.
- Control functionality varies with DAW platform — although that's hardly Audient's fault!

SUMMARY

This is a seriously impressive, competitively priced analogue console that raises the bar in terms of integrated DAW control.

Analogue Overview

The ASP2802 is a one-size-fits-all console, and what you see is what you get: there are no options or accessories. Designed to fit 19-inch racking if required, by removing the side cheeks and fitting optional rack ears (OK, make that *one* optional accessory!), this is a very modestly sized device, with a wedge-shaped chassis that rakes the control panel elegantly when placed on a flat surface.

The ASP2802's analogue circuitry, channel signal-path structure and routing facilities are very similar to those of the Zen, although it only has two auxiliary sends rather than four, and there are no subgroups — this mixer is direct to stereo only. The microphone preamps and all the channel electronics are exactly the same as those in the Zen and have a direct lineage to the original and still highly regarded ASP8024 large-format console. Each of the eight mic preamps is equipped with a button to activate phantom power, a 12dB/octave, 75Hz high-pass filter, and a polarity-inversion button, plus a 6-60dB mic gain control. All very familiar and workmanlike stuff.

Like the Zen, the ASP2802 also boasts dual signal paths through each channel strip, providing pseudo-in-line functionality. Configurable channel direct outputs to feed a DAW's multi-channel interface are also available, as are properly balanced channel insert send and return facilities.

The basic console structure comprises eight physical input channels, each with two input paths, as I mentioned (mic/line and DAW) feeding two stereo mix buses: the main stereo output and a separate stereo 'cue' bus. The primary channel path signal is selected from mic/line or DAW inputs, while the cue path takes its input either from the channel path (switchable pre-fader or post »

The rear panel features myriad connectors for different inputs and outputs. In total, the console offers an impressive 32 inputs on mixdown!

» fader) or from whichever input signal source is not routed through the channel (mic/line or DAW). The cue signal always routes to the dedicated cue bus via a rotary fader and pan control, but a Cue Assign button in the master control section can be used to route the stereo cue mix bus back on to the main stereo mix bus to expand the number of inputs for mixdown.

The cue bus is intended to feed the artist headphone monitoring, of course, and in the master section an array of buttons can be used to supplement the cue mix bus with feeds from the two aux buses, the control-room monitoring selector, two external stereo DAW inputs (DAW Mix and DAW F/B), the mix of four stereo external inputs and, of course, the built-in talkback mic (an external talkback mic socket is also provided, complete with phantom power). Comprehensive is a word that readily springs to mind when you examine any aspect of this remarkable little desk! The provision of the two external DAW inputs is intended to allow a monitor mix (DAW F/B) to be constructed within the DAW and output to the console using a pair of spare interface outputs, independently of the main stereo DAW mix, which can then be further enhanced or expanded with some zero-latency direct feeds using the console's aux or cue bus signal.

The two-channel aux sends are always derived from the channel path, and the pre-post switching for each send is performed globally with a button in the master section. At the bottom of each channel strip is a mix assign button and pan control to route the channel signal to the main stereo mix bus. On the fader strip, three large illuminated buttons provide solo, cut and select functions (more later), and the fader is a 100mm touch-sensitive motorised Alps unit — although this is used entirely as a control fader; no audio passes through it.

Unlike the Zen console, where the audio was always routed via the faders, the ASP2802 employs very high-quality VCAs (THAT's chips, in fact), which are controlled by the touch-sensitive moving faders. This is the only cost-effective and practical way of providing the switchable functionality that allows the faders to control either the analogue signal path or the DAW's virtual channels via HUI. One minor point here is that when controlling the analogue channels, the fader provides 10dB of gain above unity and the panel legending is calibrated accordingly. However, when in HUI mode controlling a DAW's virtual faders, the amount of gain above unity is determined by the DAW program itself — in Logic it is 6dB and in Pro Tools it is 12dB, for example, which may cause some confusion if you try to set levels using the fader-plate scale. This is clearly outside Audient's control, and highlights just how non-standardised different DAWs and HUI implementations still are!

The illuminated Cut and Solo buttons above the faders can affect either the DAW or the analogue signal paths, and a central assign button (labelled DAW) determines which. The Solo button can also be configured to serve as PFL, stereo AFL, Solo-in-Place or Solo-in-Front via the master section controls. The Select button function can also apply to either the analogue signal paths or the DAW, and that is determined by some more buttons in the master section of the desk, with no fewer than seven different options! Three of these affect only the analogue console operation: SiP Safe, Auto Safe, and Unity. The first isolates channels to prevent them being muted when other channels are soloed (such as reverb returns), while the second does the same to isolate channels from fader automation data from the DAW. The third option provides a quick and accurate method of restoring the channel fader to unity gain, which is useful when you want to sum stems at unity gain from the DAW, or just set the desk up to a known starting configuration.

Three other select modes relate only to controlling a DAW, and include the self-explanatory Channel Select and Record Enable modes. The Group button is used to group channels together in a DAW, while the last button is labelled Auto Mode and affects both the analogue console and DAW control, establishing the required fader automation modes (read, touch, latch and so on). The analogue fader automation is not internal: it relies on the DAW's built-in automation system, using 'dummy' audio channels to store the fader automation in much the same way as the Zen console.

Ins, Outs & Monitoring

If you've been paying attention, you'll know that so far we have identified 16 primary »

Alternatives

While there are several desks that combine quality analogue paths with some DAW control functions, all have different feature sets and are thus not directly comparable. Personal preferences, required DAW control functionality, workflow expectations, and analogue and digital I/O facilities will determine which product is appropriate. However, contenders include the **Allen & Heath Zed R16** and **GS R24** consoles (which don't have moving faders) at one end of the range, and **SSL's Matrix** — as well as their flagship **Duality** console — at the other, not forgetting **Audient's** own **Zen** console, which I've discussed in this review.

JUNO-Gi

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» inputs to the 2802 console: eight mic/line and eight DAW inputs via the channels and cues. I also mentioned in passing that there are four additional, fixed-level stereo mix inputs (via a rear panel D-sub connector), which are summed together and controlled via a rotary fader, mono switch and balance control in the master section. These summed inputs can be routed as a composite stereo signal to either the mix or cue buses (or both). There are also two independent stereo effect return inputs with all the same fader, mono and routing facilities. That lot adds up to 28 inputs that can be routed to the main stereo mix — which is evidently how the console's name was derived — and is a pretty impressive total for a compact console like this.

However, it's not actually an accurate description of the console's facilities, because there is also provision to route a separate stereo external monitor input (labelled DAW Mix) to the stereo mix bus, and the main mix bus insert point can also be configured to provide yet another mix input, too... So my rusty schoolboy maths suggests that there's a grand total of 32 physical inputs to the stereo mix! That's seriously impressive for a compact 'eight channel' desk.

The channel path has the same comprehensive Direct Output switching options as the Zen console, and the channel bar-graph meters can be flipped globally to reveal the channel signal levels or the direct output levels. A pair of push-buttons provides four options for the direct output signal routing: direct from the mic/line input stage; post the DAW selector switch and level-trim control (but pre-insert); post-insert point; or after the main fader and cut switch. These are well thought-out options that provide surprising flexibility, allowing an ultra-short 'mic preamp to DAW' mode, or enabling the use of an outboard compressor-limiter for compressing 'to tape', or even to allow fader-riding while recording — and probably several other alternative and useful ways of working too!

The Dim and Cut levels can be specified in the monitoring section, just above the in-built bus compressor.

On the output side of the desk, the built-in stereo bus compressor has a mix control that allows instant parallel compression modes, and its input and output can be accessed separately from the rear panel, to enable it to be reallocated to other signal paths if required — a very flexible idea borrowed, once again, directly from the Zen console, along with the ability to use the mix-bus insert as a parallel mix input, as I mentioned above. The master stereo fader is calibrated with unity at the top. This is another 100mm Alps fader — but is not motorised this time — and it passes real audio to the analogue outputs, rather than DC control voltages to a VCA.

The monitoring section is another typically comprehensive Audient design, with buttons to audition the main mix output or a choice of dedicated monitoring sources including the two aux buses, an 'i-Jack' (mini-jack) input, an external balanced input, or the dedicated DAW mix and F/B inputs. Stereo bar-graph meters sit across the monitor section, and additional facilities are provided for mono, polarity reverse, cut left and cut right, level dim and switching between main and 'alt' speakers. There is also an engineer's headphone output, which is switchable between the control-room monitoring source and the cue source. Although the cue output can audition the control-room source, there »

Interfacing With Your DAW

Although the ASP2802 can be used to control your DAW, it includes no A-D conversion options, and there are no USB or Firewire interfaces — so it can't serve as a computer interface itself: you will need to budget for a multi-channel audio interface of some kind to link the console with the inputs and outputs of your DAW. However, this approach is perfectly sensible, given the target market, as it enables the user to specify their own preferred audio interface, and affords the option of upgrading it later as requirements and budgets allow. I suspect that's a courtesy that many will appreciate.

NEW 8260A
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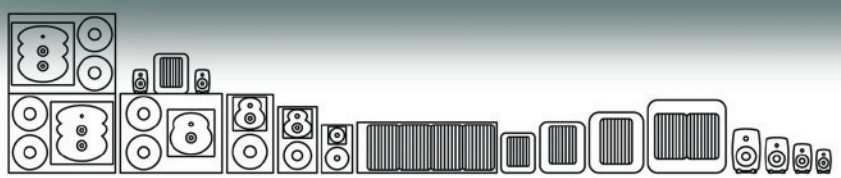

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» is no facility to audition the cue outputs in the control room, other than via the headphones. The monitoring is automatically dimmed when the talkback is active.

Almost all of the audio I/O on the 2802 is balanced, using XLRs for the channel mic inputs, external talkback mic input, mix, cue, and aux outputs, main and 'alt' speaker outputs and the external monitoring inputs (DAW mix, DAW F/B, and external). All of the channel and mix inserts are on pairs of quarter-inch TRS sockets, with additional balanced jack sockets for the compressor I/O, the stereo effect returns and the channel line inputs. Three 25-pin D-sub connectors cater for the channel direct outputs, channel DAW inputs, and the four stereo summing bus inputs. Mains power is via the usual fused IEC connector, with an adjacent power switch, and the link to the DAW host is via a standard RJ45 Ethernet (10/100T) port.

DAW Control

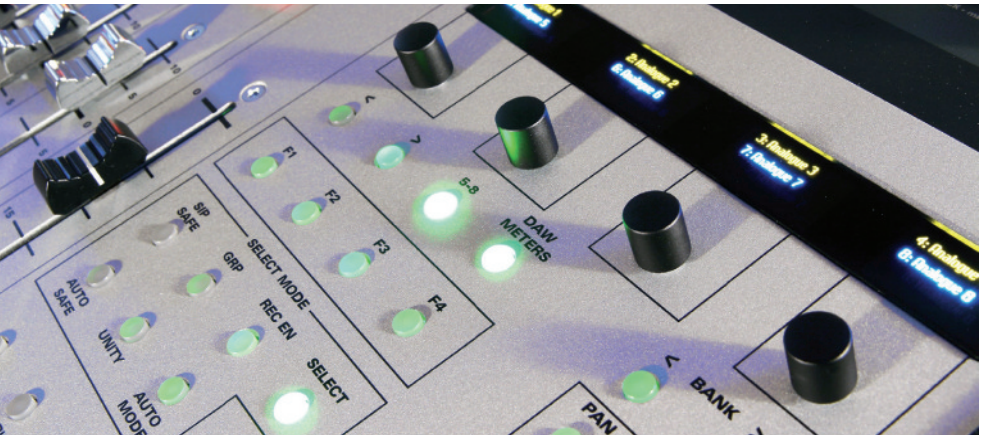
I was surprised to learn that the DAW interface is via a standard Ethernet (Cat-5e) connection, rather than the more usual MIDI interface, but it is a more modern and effective way of working. Bespoke utility driver software called AuNet is provided to form the bridge between the DAW's remote control interface and the console's Ethernet data stream, and this software will run on either PC or Mac platforms, and in 32- or 64-bit environments.

Essentially, the AuNet driver creates four virtual MIDI ports, with one pair conveying the control-surface data and the other pair handling the fader automation — each pair associated with a separate HUI interface. The desk can be connected via a standard Ethernet router, and the default 192.168.0.1 IP address can also be changed manually or automatically using DHCP. Configuring the console is actually very straightforward and takes only a few moments, confirming the ATA connection and then selecting the appropriate DAW settings. The data handling inside the desk is courtesy of an ARM7 processor, which I'm told has plenty of capacity for future expansion opportunities.

The 2802 console is designed to appear as a standard HUI interface to the DAW software, and the console has built-in options to optimise the configuration for Pro Tools, Logic, Cubase and Nuendo — although other DAW configurations can be added through future firmware updates, passed easily from the host computer via the network connection. Updates will be downloadable from the Audient web site.

The bulk of the console's master section is given over to the DAW control facilities, with a full set of large, illuminated transport controls and a jog/shuttle wheel at the bottom. Additional buttons are included for cycle, marker, nudge and scrub functions, along with a standard set of up-down, left-right navigation buttons. There is also a Shift button, a setup button (for configuring the system) and a DAW button that toggles the control-surface functionality — such as the faders and mute buttons — between controlling the DAW and controlling the analogue circuitry.

I've already described the seven select-mode buttons, but above these are four programmable function keys, and above those is a pair of page left and right buttons that accesses additional control parameters for the function currently assigned to the rotary encoders. There's also a button to reallocate the four channel encoders and OLED displays to control channels 5-8 of the currently selected eight-channel subset, and a button to flip the channel bar-graph meters to show the DAW channel levels instead of the analogue channel levels. Over to the right is an additional pair of buttons to scroll channels in banks of eight, and an additional quartet of buttons to select the current role of the encoders, with options for Insert (accesses DAW channel plug-in controls), Pan, Aux and Assign (channel I/O routing). Above all these are the four rotary encoders (which double as switches) and their associated colour OLED displays.



As well as rotary encoders for making pan settings, there are four 'function' buttons, which can be used to control features on your DAW.

Of course, the complexity here is that the precise functions of each of these buttons and encoders varies to some extent, depending on the capabilities and configuration of the DAW in use, so for simplicity I'll describe them only in terms of working with Logic. The basic HUI control functionality is the same for all supported DAWs — but some ancillary functions vary or are absent entirely.

As I mentioned earlier, the console's fader and mute automation for the analogue signal paths relies on the DAW's automation features, and Audient's superb collection of DAW-specific manuals detail how to create the required eight dummy tracks. Most DAWs provide a comprehensive set of automation modes, and Logic offers Touch — probably the most useful — Latch, Write and Read options. One possible source of confusion is that the console powers up with automation safe mode automatically selected on all channels — a sensible precaution, perhaps, but it requires the user to manually deselect auto-safe for each channel as required. The Sel button above each fader is green when Auto-safe is engaged, and white when automation is enabled, so the status is very obvious once you're familiar with the console.

When working with Logic, the cycle and marker buttons do exactly what you would expect, but the Nudge button is currently unsupported in Logic, and so does nothing at all. The navigation controls move around a Logic project as you'd expect, providing track-select and region-select modes, and when used with the Shift button, they serve to alter the horizontal and vertical track-display zooms. To scroll around in a large project, the bank buttons

To the right of the faders is a well-featured DAW transport-control section.

move in blocks of eight channels, while using Shift with the bank buttons moves in single-channel increments. The first four channels in the selected block are detailed in the OLED displays, with the last four being accessed via the 5-8 button.

The OLED displays also indicate which parameters can be adjusted with the encoder, as selected with the pan, aux, insert and I/O assign buttons — and if there is more than one parameter (as in the case of the various aux sends), the page left/right buttons allow access to them. One function that worried me momentarily, because I thought there was something wrong, is that the OLED backlights dim automatically after a period of inactivity. As soon as you start using the console, they illuminate to full brightness again, which makes perfect sense — but it is disturbing if you're not expecting it, or if you come back to a console where you can't read the displays anymore!

There are also some 'hidden' functions to speed workflow, such as double-pressing the aux button to flip the aux-send levels from the four encoders onto the eight channel faders — making it much quicker and easier to set up monitor mixes or reverb sends, for example. The aux button flashes to warn when this mode is active. Similarly, pressing the assign and aux buttons together provides a quick way of accessing the aux send buses to create new sends or re-route existing ones.

When working with Logic, the four assignable F-keys are configured by default to clear overload flags and select three alternative screen sets — although these functions can be changed in the DAW if required. (If you're working with Cubase or Nuendo, the function keys currently have no function.)

Conclusion

Although inherently complex and involved, DAW control using the ASP2802's interface is actually very logical and intuitive in practice — it's certainly no more difficult than using any other HUI-based hardware controller — and Audient's dedicated manual sections covering the different DAWs make learning and configuring the functionality very straightforward, without you having to wade through the irrelevancies of other DAW systems. The OLED displays are quite superb, and the controls are all very tactile and feel solid and reliable.

In fact, the whole thing feels really professional and workmanlike, and frankly I'm amazed Audient have been able to achieve so much for such a modest cost. The audio quality is beyond reproach, and the facilities are well thought-out and remarkably versatile. Whichever way you look at it, the ASP2802 represents seriously good value for money to anyone wanting to add first-class front end and monitoring facilities to an existing DAW setup, while gaining elegant and well-integrated hardware control. Other manufacturers have a new benchmark to aspire to!

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Mackie Onyx Blackjack

USB Audio Interface

Mackie's Blackjack is the smallest interface yet to feature their acclaimed Onyx mic preamps.

PAUL WHITE

The Mackie Onyx Blackjack is a combined USB audio interface and monitor-level controller aimed at those who only need to record one or two inputs at a time, but who want the preamp and digital conversion quality of Mackie's larger Onyx interfaces and mixers.

The unit is designed for desktop use, and Mackie's designers have clearly gone for simplicity, quality and very robust packaging. The folded-steel case has a built-in rear leg so that it sits at a sensible angle on the desk, and all the connections are kept at the back, out of the way.

The Blackjack works with both Mac and PC computers and is compatible with all the major DAW software, other than those software products tied to the manufacturer's own hardware. While the Blackjack is class compliant and will work without the need for drivers, PC users running Windows are advised to download and install the latest ASIO driver from www.mackie.com/products/onyxblackjack for the best performance.

The Blackjack comes with a bundled copy of Mackie's Tracktion 3 software, which provides a pretty painless way to get into computer recording if you're not already wedded to a particular DAW.

Ins & Outs

The Blackjack features two Onyx preamps, having mic/line inputs on 'combi' jacks/XLRs, with individual DI switching for using the line jacks as high-impedance inputs, and switchable global phantom power. The monitor outputs are on balanced quarter-inch jacks, governed by the Monitor level control, and there's a headphone output and accompanying level control on the front panel.

The only thing missing is an S/PDIF digital input; you may not need it very often, but they come in handy for importing DAT tapes or other digital formats.

Features

The Onyx mic preamps, which have simple metering in the form of a dual-colour signal/overload LED, are the same as those used in Mackie's Onyx-series mixers, and our reviews of these products have always included favourable comments about their sound. There is no pad or low-cut filter on this version but there's plenty of headroom, and impressive signal transparency with up to 60dB of gain. Mackie claim their Onyx preamps rival the most expensive boutique mic preamps for quality. While we probably wouldn't go quite that far, they certainly are impressive.

Conversion is via highly specified Cirrus Logic chips, and power for the circuitry comes from the USB port, so there's no need for a separate PSU. These 24-bit Cirrus Logic converters are capable of a 114dB dynamic range (A-weighted), which is way beyond the signal-to-noise spec of most studio gear. Importantly, the gain staging within the unit means that the preamp circuitry is correctly matched to the A-D conversion, to maintain headroom while minimising noise and distortion.

An important feature is the Input Monitor knob, which mixes the input signal with the DAW return mix for simple latency-free 'source' monitoring. This means that you can set larger buffer sizes when recording without suffering that annoying delay in the headphones. Just remember to turn software monitoring off in your DAW to avoid hearing direct and delayed sound at the same time! A Mono button puts the input-monitored signal into

Mackie Onyx Blackjack
£210

PROS
- Built from steel, just like a Mackie mixer.
- Good audio quality.
- Tracktion 3 software included.

CONS
- No digital I/O.

SUMMARY
The Blackjack is an irresistible combination of chunky-but-compact hardware with exactly the right facilities for basic recording to a high standard.

both left and right phones and speakers if preferred; otherwise channel one is heard on the left and channel two on the right.

In general, the technical spec is similar to what you'd expect from a Mackie console, except, of course, that the frequency response is dictated to some extent by the sample rate used. You can run the Blackjack at sampling rates of up to 96kHz, and it delivers a 15Hz to 22kHz response accurate to 0.2dB. At 44.1kHz, you still get a 15Hz to 21kHz range. The frequency response of the preamps and other analogue components extends some way beyond this, with the preamps being only 0.2dB down at 50kHz. The converter dynamic range is inevitably greater than the noise floor of the circuitry, but again the circuitry performs pretty much as you'd expect from a Mackie mixer, with a generous input headroom up to +10dB in mic mode and +25dB for the line inputs.

At the rear, we find a USB connection, a pair of monitor outputs on quarter-inch jacks and the two 'combi' jack/XLR inputs.

Testing

Plugging the Onyx into my Macbook Pro, I launched Logic Pro and immediately found the Blackjack available as a named input source — no drivers, no fuss. As with most mic preamps that use standard potentiometers, the useful mic gain does tend to be bunched into the last one third of the gain control's travel, but that didn't present any problem, as I set the gain for a speech test at a 96kHz sample rate. The playback quality was excellent, so I repeated the test at 44.1kHz. There was a small but discernible difference between high and low sample rates, the higher rates seeming to give a fuller, slightly smoother sound that some might describe as more 'analogue'. The difference is quite subtle, and is in line with what I've heard from other mid-priced interfaces, so it is down to the individual to decide whether the extra fidelity is worth trading against half your computer's track playback and plug-in count. I can't really ascribe a specific tonal character to the Onyx preamps — they just sound clean and musical to my ears.

Alternatives

There are many similarly sized interfaces available, including the less costly **Presonus Inspire** (which is Firewire-based), but nothing I've used yet has matched this combination of a good-quality interface with practical monitoring in such a robust package.

Music playback through the Blackjack was also to a high standard, even at 'normal' sample rates, with a full but well-controlled low end and smooth, extended highs. The headphone output delivers plenty of clean level, too.

Conclusion

The bottom line is that this simple and very affordable little unit delivers both on sound quality and convenience, in a package that's as robust as you could wish for. I had no problems with USB whining in the background, and unlike some of the lighter plastic interfaces on the market, it didn't fall off the desk as soon as I plugged in a mic cable!

Other than an S/PDIF input, there's really nothing more you could ask for in a basic interface, and while there are cheaper alternatives, once you add in the cost of a monitor controller/headphone amp and take into account both the mechanical and audio quality, the Blackjack starts to look like a real bargain. ■■■

£ £210 including VAT.
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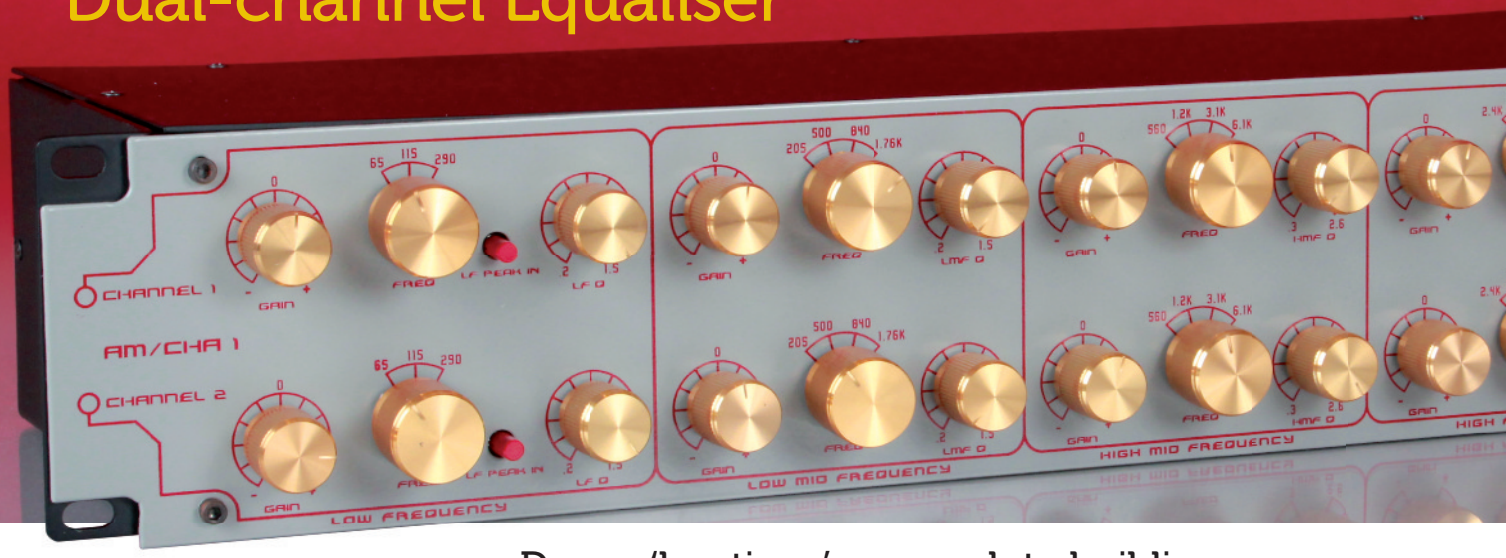
Black Lion AM/CHA1

Dual-channel Equaliser



HUGH ROBJOHNS

Does a 'boutique' approach to building outboard have to mean inflated prices? Not according to this manufacturer...

Black Lion are a Chicago-based company that started out developing and providing upgrades for other manufacturers' equipment — generally addressing the technical corners often cut in trying to bring a product to market at a budget price. The company's approach is that great-sounding equipment doesn't necessarily have to be expensive, it just needs to be designed properly — and they are right about that, in my view.

After developing a good reputation and loyal following for their numerous product upgrades, Black Lion started to branch out and develop their own boutique products from scratch, with the declared aim of producing "excellent products at exceptional prices." They're not mass producers, either: everything is done in house, and mostly by hand.

One of the company's latest in-house designs is a two-channel, transformer-coupled and inductor-based passive equaliser (but with active gain-makeup stages), called (rather mysteriously) the AM/CHA1.

Overview

This 2U-high, rackmounting unit, which extends roughly 145mm behind the rack ears, is all very old-school in concept and technology, but thoroughly modernised in construction with the use of surface-mount, integrated-circuit gain stages and supporting circuitry.

Of course, plenty of manufacturers have inductor-based equalisers in their catalogues, from the classic Pultec and Neve designs (and their numerous clones), through to more modern API and Massenburg products — and many more besides, with most being very expensive, too. However, the Black Lion AM/CHA1 was designed from the outset to be an original design, rather than a clone of something else, and to be more cost-effective than is, perhaps, the norm for inductor-based semi-passive units. However, before you rush off to smash the piggy bank and count the penny pieces, this is still an expensive device in real terms — but it does represent good value for money for a product of this kind.

The AM/CHA1's circuitry is actually based on a standard state-variable filter topology, but uses carefully optimised, custom-wound Cinemag inductors in the filter circuits, and Edcor PC-mounting steel-cored transformers for both the input and output stages. These low-loss coupling transformers use grain-oriented iron-silicon alloy laminations, with a 10k/10kΩ-impedance type handling the input and a 600/600Ω impedance

Black Lion AM/CHA1 $1195

PROS

- Inductor-based equalisers have a recognisable and attractive sound quality.
- Transformer-coupled I/O adds to the vintage character.
- Four bands with well-chosen frequencies and overlaps.

CONS

- Review unit chassis not earthed!
- No centre detents on gain controls.
- Hard to see whether the push buttons are in or out.
- Unsecured power LED.
- No manual supplied and none available on the web site.

SUMMARY

An interesting inductor-based equaliser with all the sonic virtues that approach bestows, supplemented with some old-school transformers for even more vintage flavour. This is an impressive product at an attractive price — although I'd pay a little extra for centre-detents on the gain controls and band bypass buttons!

for the output. Amorphous-core Lundahl transformers are available as a cost option, if preferred. With this amount of iron in the signal path, it will come as no surprise to learn that the AM/CHA1 has something of a vintage sound character, with distinct warmth and sheen properties! That character is largely maintained even when the EQ is bypassed, which is useful if you want to add a bit of flavour to a mix.

The rear panel carries the IEC power socket with an integral fuse holder and voltage selector, while four quarter-inch TRS sockets handle the audio in and out. The front panel is divided into four equaliser-band sections, each with a rotary switch to select the filter frequency, and continuously variable Q (bandwidth) and gain controls (the latter without a centre detent). A miniature toggle switch is used to power the unit, and two push-buttons provide independent channel bypasses to complete the facilities. The review unit had a hole where the power LED was supposed to be (it had fallen back inside because it is only held in place by the goodwill of its connecting wires!), but there are no other lamps or illuminated buttons to indicate when the unit is working or bypassed (and the push-button status isn't at all obvious).

The frequency ranges are switched rather than continuously variable, because of the need to insert different inductors to adjust the filter tuning, of course, but the four bands do overlap quite generously, to maximise usability. The top three bands all have four frequency options, while the lowest band has just three — giving 15 frequency options altogether. For those with a passion for the numbers, they are: Low band, 65Hz, 115Hz and 290Hz; Low-mid band, 205Hz, 500Hz, 840Hz and 1.76kHz; High-mid band, 560Hz, 1.2kHz, 3.1kHz and 6.1kHz; and High band, 2.4kHz, 3.7kHz, 8.2kHz and 15.6kHz. The top and bottom bands can also be switched from shelving to a peaking response, via small front-panel push-buttons. The variable Q of the two lower bands can be adjusted over a range of 0.2 (nearly five octaves) and 1.5 (just under one octave), while the upper mid spans 0.3 (nearly four octaves) to 2.6 (half an octave), and the top band 0.3 to 3 (just under half an octave).

The review unit arrived configured for 110V mains power and was not marked to confirm its CE compliance. Although the voltage selector integrated into the IEC power socket made changing the supply voltage easy enough, I decided to perform a quick PAT safety inspection before powering the unit up (I've been burned — once, quite literally — by uncertified equipment before!). Unfortunately, a safety earth continuity check revealed that the earth terminal of the mains inlet socket was not connected to the chassis metalwork at all, despite this being a fundamental requirement for a Class 1 device. An internal inspection revealed that the mains earth went to the audio circuitry ground, but nowhere else!

Audio Precision Test Plots

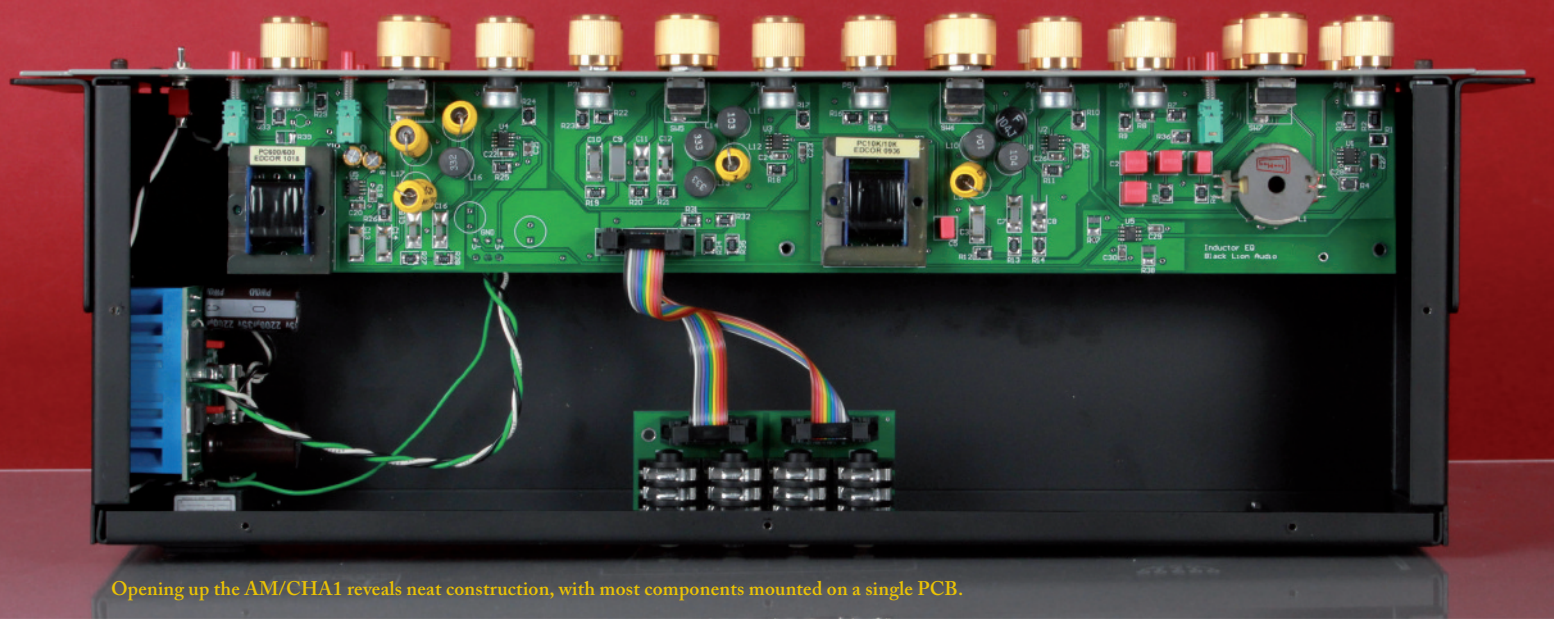
In the course of reviewing this device, Hugh used an Audio Precision Analyser to make some technical measurements. You can find the resulting charts on our web site at www.soundonsound.com/sos/dec10/articles/blacklionamcha1media.htm.

Consequently, all my further testing of the unit was carried out while it was powered from a mains isolation transformer, to ensure my safety. This interesting approach to mains-power safety does need attention before the unit can gain CE certification and be deemed acceptable for use in Europe, so I raised this directly with the manufacturer and was told: "When we were testing the unit, for some reason we kept on finding that it was quieter [*when grounded*] that way, and the grounding was still reliable in the case of a short or over-voltage, so we didn't pay it too much attention. The difference, however was not enough for us to keep it that way, so in the end we have actually decided to go back to grounding directly to the case, because as we see it (and I'm sure you agree), quieter or not, it's just a more reliable grounding point." So presumably future retail models will have properly grounded casework, but this might be worth checking before purchase.

Getting Technical

Running through a standard set of test measurements, the overall technical performance was very good, although it became clear that the gain through the unit is affected by the loading impedance — as is common with transformer-coupled devices. With a 600Ω termination on the output, the overall gain was about 3dB, but that rose to 5.5dB when feeding a high-impedance destination. This is not of huge practical significance, perhaps, but worth bearing in mind when optimising a system's gain structure. It was also apparent that the »

The rear panel of the AM/CHA1 includes inputs and outputs on balanced TRS jacks.



Opening up the AM/CHA1 reveals neat construction, with most components mounted on a single PCB.

» dominant third-harmonic distortion on the unit's output was attributable to the I/O transformers, and remained even when the EQ was bypassed.

The variety and flexibility of the AM/CHA1's equalisation curves is comprehensive. As is fairly common with this kind of equaliser, substantially higher levels of boost and cut are available when lower Q settings (narrower bandwidth are employed. Roughly ±16dB of gain range was available for the lowest Q-value settings, rising to a ±26dB swing with the highest Q setting. The unit has a headroom margin of +22dBu (rising to +24dBu in bypass), so great care is needed when using high boost levels on hot input signals with high Q settings. This wide variation in maximum gain is because of the 'constant energy' nature of the passive filter arrangement used here — you can see exactly the same kind of thing in a Pultec EQ, for example. Since the total energy within the filter band is always the same, if the bandwidth is reduced the amplitude increases, and vice versa. Most modern active equalisers — such as those in most consoles — don't tend to use constant-energy filter designs, so this effect is not that common. Given that this equaliser is best suited to subtle tonal sculpting, the gain range, even in low-Q modes, does seem excessive, and perhaps a reduced range would provide more resolution and control. I rarely found the need to dial in more than about 6-8dB of gain or cut during my trials, for example.

One oddity that emerged from my Audio Precision testing was a strange frequency-response distortion effect only apparent in the upper-mid band. I suspect that this was due to inductive cross-coupling between channels, and it only occurred when one channel was on maximum boost and the other on maximum cut — so it's very unlikely to be a problem in practice, and no other bands were affected in the same way.

Again, I raised this minor issue with Black Lion and they said: "Small quirks like that can often times be the result... of the varying tolerances of inductors [*and it*] appears that it's not actually cross-coupling with another band, but a sudden jump in the filter's reactance. Luckily that's not an issue that would cause the filter to actually become unstable. Thanks for the heads-up on that point. Definitely gonna take care of that." I find that a most reassuring response from the manufacturer.

In Practice

The absence of centre detents on the filter-band gain controls makes zeroing the EQ more difficult than it should be, but the controls are nicely weighted, feel solid, and are arranged in a nice, spacious panel layout. Despite the absence of any manual (in the box or on the web site), the device is perfectly intuitive to understand and use. I found the way the gain changes quite dramatically when altering the Q control to be a little frustrating occasionally, as it made EQ optimising a rather iterative process, but the AM/CHA1 is not alone in this operational quirk.

The frequency options available in each band are well chosen and seem musically appropriate, and the unit works very well when it comes to gentle, creative tonal shaping. This isn't a surgical tool for removing specific unwanted signals, but does work well to pull out the best, or to recess any irritating aspects of a mix or source's tonality. For this reason, a more moderate gain range for each section would allow the EQ to be applied with a little more finesse and control, but perhaps this is nitpicking.

Verdict

The AM/CHA1 definitely brings its own distinct sonic character to the party, with a musically benign but noticeable warmth and thickening of the bottom end, and a smooth, almost creamy high end. Good analogue equalisers like this are a joy to use, with minimal settings often delivering sublime enhancement. The top 15kHz filter setting in peak mode with just a hint of gain is great for adding air and sparkle, for example, and overall the AM/CHA1 is a delightful-sounding equaliser. A mastering version with switched and matched controls throughout would be a big hit, I'm sure. ■■■

Alternatives

Inductor-based equalisers at the AM/CHA1's price are few and far between, but a similarly priced and similarly musical alternative would be the **JDK R24**, which also features input and output transformers.

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MIKE VERNON

Producing The Blues

Mike Vernon produced some of the greatest blues records of all time. A full decade after retiring, he's back in the studio with some of the British blues scene's brightest lights.

Mike Vernon (right) with his brother and sometime studio partner Richard Vernon.

MATT FROST

Mention the name Mike Vernon to any self-respecting blues fan, and you can guarantee that it won't be long before said fan is reeling off the names of classic records he made as a producer during the late-'60s British blues boom. As well as manning the helm for many of John Mayall's recordings — including the groundbreaking 1966 album *Bluesbreakers With Eric Clapton* — Vernon produced numerous other Brit blues artists including Chicken Shack, Peter Green, Fleetwood Mac and Ten Years After, and US blues stars including Otis Spann, Champion Jack Dupree and Eddie Boyd also recorded albums with Mike, for his legendary Blue Horizon label. Although he'll probably be best remembered for blues records, Mike Vernon's production endeavours have also included artists as varied as David Bowie, Focus, Level 42, the Proclaimers, Bloodstone and Dr Feelgood, as well as his own groups, the Olympic Runners and Rocky Sharpe and the Replays.

In 2000, having become disenchanted with the state of the mainstream record industry, Mike decided to call time on his career and moved to the Spanish countryside, limiting his musical activities to occasional live events in his Spanish locality and Blue Horizon reissues. A few years ago, however, Mike suddenly decided that he rather fancied getting back into the studio again, after a bit of gentle prompting from an old blues player friend of his.

New Horizons

"We have a big blues festival here that's been running here for about 20 years, and two years ago a very old friend of mine, Sherman Robertson, played," Mike explains over the phone from Spain. "And he played because I got him the gig. He was absolutely, outrageously brilliant and, of course, we got talking afterwards and I kept saying, 'It's about time you made another studio record.' He said, 'Well, I'm ready, I'm ready… but I'm only going to let one person produce it, and that's you.' So I said, 'Great, let's get some songs together, let's find a backer and let's get the whole thing sorted!' I spent about five or six months working on it, and then suddenly Sherman turned round and said, 'I don't want to do it!' I think the phobia kicked in, and I was left feeling very deflated, because I was really actually looking forward to going back into

the studio one more time and just seeing if I still liked it!"

Mike Vernon wouldn't have to wait too long to find out. Thomas Ruf, head honcho of top German blues label Ruf Records and backer for the Sherman Robertson record, had other artists on his roster he thought would benefit from Vernon's legendary expertise. Enter two talented young British blues guitarists and singer-songwriters, Dani Wilde and Oli Brown, the musicians who can duly lay claim to having brought Mike Vernon out of retirement.

"Thomas came to me and said, 'I need to make a new album with Dani Wilde. How do you fancy that?'" recalls Vernon, "Well, I had to admit I'd never heard of her, so I said, 'Send me what there is and her new songs,' and I liked what I heard. And then I got another call from Thomas saying, 'Right, let's just put Dani on hold… I've been left in the lurch. The producer who was going to do Oli Brown's new album has pulled out, would you like to do that?' And I again had to admit, 'Sorry, never heard of him!' Anyway, Thomas sent me Oli's stuff and I thought, 'Oh God, this kid's actually fantastic! I have to be involved in this.' And so I ended up making two records in the space of three months, which wasn't as I would have planned, but I don't regret it."

One thing that really impressed Mike Vernon about both Oli and Dani was how easy they were to work with. The cockiness and arrogance of certain bands and artists during the '90s had been one of the reasons Mike had wanted to get out of the business in the first place. "I have had the misfortune in the past to work with people who are just so big-headed and who won't take any advice at all, and that's very irritating!", laughs Mike Vernon. "But Oli was such a nice kid, was really, really into the music and wasn't like 'Well, I'm the new kid on the block, don't try and tell me what to do!'… and I have to say the same about Dani. I want commitment from people. I give 100 percent myself — I always did — and I feel quite aggrieved when the artist themselves spends more time rolling joints and getting drunk and turning up late. I just don't need all that! I got a really professional approach from both Oli and Dani, which was much, much appreciated, and I think that actually shows in the records."

A Very Live Sound

Oli Brown's *Heads I Win, Tails You Lose* and Dani Wilde's *Shine* were both recorded at Platform Studio in Hurst, Berkshire, and were engineered and mixed by Sean Lynch and Damon Sawyer respectively. Mike Vernon's approach to both recordings echoed the way he's always favoured producing bands in the studio: getting things sounding as live as possible.

Both of Mike Vernon's recent projects were recorded at Platform Studio in Berkshire.

"If you listen to *Bluesbreakers…* and a lot of the Fleetwood Mac stuff that I was involved with, it has a very live sound because most of it was recorded all at once," explains Vernon. "And we tried very hard to do that with Oli, although we didn't quite achieve it, but the thing is that I think in terms of the overall sound, it does sound pretty big and like they are actually all playing at once, which pretty much they were. I've always tried to do it that way. We pretty much did Dani's record the same way really — we had her play and sing live with the band, but she redid most of the vocals, so that she could really concentrate on what she was doing. But the fact that she was singing live when they were actually playing the track made an enormous difference to the feel of what was being played, and that's exactly what happened with Oli. Oli was in the control room with Gary the bass player and Oli was singing into the mic, and Jamie was drumming in the studio through the glass, and they were all playing to the nuances of a live vocal."

Working with the right engineer is always a key prerequisite for Mike Vernon when he's in the production hot-seat.

"Given the choice, I try to find people that I know, who I can work with, who are of a similar nature, who have similar ears and like what I'm going to be doing," he says. "I don't want to have an engineer working with me who doesn't like listening to 12-bar blues all day, because that would be really tedious for them and that's not good for anybody. I think that »

Photo: Eckie

Oli Brown.

» engineers and producers have to have that sort of relationship."

When it comes to mic selection, Mike Vernon has never been too concerned. He tends to let his engineers make their own informed choices.

"There are lots of very good microphones around, but of course they can be inordinately expensive!", says Mike. "I mean we'd all like to be able to have one of those great big RCA ribbons but you can't get them for love nor money and if you could, you'd be paying another mortgage out! But I don't know if the difference it creates is sufficient enough for, you know, the average punter. I do know people who can tell the difference and can say, 'Well that sounds to me like a Neumann…' which is extraordinary, really, but I'm not one of those people. I'm more of a feel merchant: if I don't have the right kind of feel from the band, the right kind of approach from everybody involved, then it's not much good to me. It won't make any difference what microphones I use, it won't sound any better!"

Both albums were mixed using Cubase and each record was completed in a refreshingly short amount of time.

"These records were made in 10 days and I would prefer to do that than spending three or four weeks," explains Vernon. "I'm not really into all that. I've had to do it before and I don't want to have to do it again. That's the only part of it that I don't enjoy, because it all takes too damn long! I like the old days, when John Mayall would make an album in four days!"

Old School Methods

Unsurprisingly, for someone who places the highest importance on capturing a band's performance as live as possible, Mike Vernon prefers to work in the analogue domain when he can. "I'm a great fan of analogue, always have been, and spent the greater part of my career working on analogue. For me, analogue makes you focus more on what you're doing. You have to actually give the performance of your life to be able to get the result. When you work digitally, you have so much more scope, and if you make a mistake, it can be corrected at the drop of a mouse. It's just click here, copy and paste it and it sticks. I'm not saying I don't subscribe to it, because I do and I have done. In fact, just recently, within the last couple of months, I've been involved in a project here in Spain, which is another blues project, with Louisiana harmonica player Lazy Lester. The whole thing was recorded digitally using Pro Tools. When you've got a guy who's flying in and a band who are flying in, and they're only here five minutes and then they're gone, you don't have time to review everything and you suddenly find yourself copying four bars and pasting it in and things like that. You might have been able to do that in analogue days, and I have been known to do it, but it's not what you normally do is it? What you would ideally do is say, 'It's crap, lads, you've got to do it again!' But the digital format does allow you a lot more room.

"The truth of the matter is that I think that about 95 percent — or maybe even slightly more — of the people that buy records these days wouldn't be able to tell the difference. If you played them an analogue version of something and then played them the digital version, they wouldn't be able to tell the difference. There's only certain nuances that elephant ears like yours truly can hear… it's the subtleties in certain sounds, and the smoothness, the warmness or the hardness of it that make it one thing or the other."

Where It All Started

Mike Vernon fell in love with music at a very early age and was soon "sponging up" all the rhythm & blues, rock & roll and blues tracks he could find. He began working for Decca Records in 1962 while he was still in his teens.

"I didn't really have [*a job description*] in those days," says Mike. "I suppose it was what you'd now call a gofer — 'Make the tea, go for this, go for that, take this up to the studio' — and that was about as far as it went. It was a stuffy old place, full of stuffy old people, and I just felt that it needed an injection. I was far too young to ever say such a thing, but I just felt that there would come a time where Decca would become part of the real world, and I'd like to think, actually, that I did have some major part in that, along with my immediate boss, Hugh Mendl, who gave me enough rope to hang myself 10 times — put it that way!"

It was Mike Vernon's obsession with the budding London blues scene that helped him develop into one of Decca's youngest record producers.

"I just took opportunities," he explains. "I was such a blues freak, and I was always out at night in London at any one of about half a dozen clubs, listening to the Yardbirds »

Photo: David Wedgbury / Decca Press Dept.

Mike with Neil Slaven and Keef Hartley, circa 1970.

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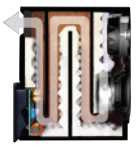

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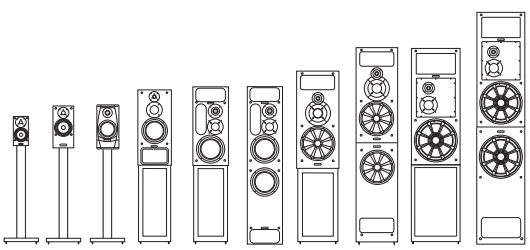

» and so forth, and that's how I got to meet Eric Clapton in the first place. I used to go see John Mayall at the Flamingo and we became known to each other and that's really how John Mayall got the renewed deal at Decca… I went to Hugh Mendl and said, 'We need to pay some attention to John Mayall's Bluesbreakers, especially now he's got this young ex-Yardbirds guitar player, Eric Clapton, who's turning the blues scene completely upside down. He's going to be a major force as a guitar player in the future. We need to nab this band while we've got the chance.' And he said, 'If you say so, go ahead and do it!' so we negotiated the deal. I got involved as producer immediately, and that was really how it all started."

Mike Vernon (third from left) has also stepped out from behind the desk as Eric Rondo of Rocky Sharpe & the Replays.

Photo courtesy Ace Records

Beano

Mike Vernon tells us about the challenges he had while recording John Mayall's classic 1966 *Bluesbreakers With Eric Clapton* LP, fondly known as the 'Beano album', due to the fact that Eric Clapton was snapped reading a copy of the famed kids' comic on the sleeve.

"The whole plan was to make that record as live-sounding as we possibly could, and in those days that was not easy, because there were so many restrictions in terms of the way people used to do things," says Mike. "Everything was always, 'Well, you must do it this way, you must do it that way, you must always have the microphone only so far from the actual cone of the amplifier and the amplifier must only be turned up to three or four for the optimum sound reproduction!'

"Clapton had said, 'This is going to be your biggest challenge, recording my sound!' We didn't realise how big a challenge it was going to be but, thank God, we had a young engineer who became a very famous producer, Gus Dudgeon, who was ready for any challenge whatsoever. Sadly, he's no longer with us [*Dudgeon died in a road accident in 2002*], but I can remember seeing his face the very first time Clapton plugged into the Marshall stack and turned it up and started playing at the sort of volume he was going to play. You could almost see Gus's eyes meet over the middle of his nose, and it was almost like he was just going to fall over from the sheer power of it all! But he dealt with it in inimitable style, and after an enormous amount of fiddling around and moving amps around, we got a sound that worked. I think all the solos, with the possible exception of 'Stepping Out', were done live. You can actually tell they were, because the drums suffer as a result of it. There was an enormous amount of guitar on the drums. The studio wasn't very big — it was big enough, but nobody had had to deal with a band making that kind of noise."

In 1968, just two years after the great success of the 'Beano' record, Mike Vernon left Decca and went independent. The move was largely a result of the fact that Vernon's cult Blue Horizon label — upon which he'd been releasing small-run blues recordings since the mid-'60s — had gained such a great reputation on the British blues scene.

"It just sort of snowballed, to the point where Peter Green was going to leave John Mayall and form his own band and he said to me, 'I want you to record our records and I want them out on Blue Horizon. I don't mind if we're with Decca, but I don't want it on any other label but Blue Horizon,'" explains Vernon. "I did the very first demos with what would become Fleetwood Mac, and they got offered to Decca, and they weren't rejected, but they wouldn't put the record out on the Blue Horizon label… so we offered it to CBS and CBS took it and took the label identity as well. But once that record came out and was something of a success, I got the dreaded phone call from the seventh floor at Decca, got called in and was told, 'You can't produce records for other record companies!' I said, 'Well, »

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The Engineer's Perspective

Damon Sawyer started Platform Studio back in 1999 and, as well as recording and mixing Dani Wilde's *Shine* album, he played the role of assistant engineer on Oli Brown's *Heads I Win, Tails You Lose*. So what was it like working with Mike Vernon?

"Mike is absolutely by far the most professional guy I've ever worked with. He's so in control of the whole project and always involved, at every point. Whenever there was a decision to be made, he was just there in the driving seat. One of the reasons I really enjoyed working with him was because he's got what I call the old-school approach, with all the emphasis on performances, and that's the way I like to work as well. I'm a player: that's what I started off doing, so I'm all for capturing the performance. And Mike's so good with the artists. The primary thing was to keep them comfortable. When we were recording Dani, if she said, 'Oh, I fancy doing some vocals,' you'd just down tools and go to vocals, whereas quite often people will go 'Hang on, we're set up for this...' With Mike, if the artist wants to sing, she sings, and I must say I learnt an awful lot from working with him."

Damon talked us through the mic placements he used on both albums.

"Oli's guitars — most of the rhythm tracks and everything — were recorded with a Prodipe ribbon mic and an SM57, and for guitar overdubs, solos, there'd be a room mic as well. I think most of the time it was probably an AKG C414 in the room. We've got quite a nice live space, which is great for drums and ambient miking. Oli did swap amps a lot. It wasn't just one straightforward rig — there were different rigs for different tones and solos, whether it be lead track or rhythm. I think there were three different amps in there. There was the Blackstar [*HT Stage 60*] and there was a Kustom [*Coupe 72*], and I think there was a Fender as well. I don't think they were actually using multiple amps at any one point, but most of the rhythm tracks were actually cut live with the drums and bass, then some solos were dubbed. The heart of both of those albums was, again, the old-school approach of actually capturing the performances with the rhythm section as it is. On Dani's, we had a Fender Blues Junior and a Super Reverb… we used different amps depending on the tone. It was a similar mic setup, with a mixture of the ribbon, a 57, and a 414 sometimes. Mike insists on having that room mic for any kind of solo overdub and it helps the solos cut through. I used the early 414, the EB, because I do think they sound quite different to the newer reissue stuff like the XLS. I always think the top end's a bit too much — it's a bit more natural with the vintage stuff. The drum setup had an AKG D112 on the kick, a Beyer M201 for snare, a pair of vintage C451s for overheads, two 414s on the two toms, actually set to hypercardioid, and I also use a home-made 'sub kick', which is actually a 15-inch speaker, but it's great — and there was a room mic as well, a Prodipe valve lamp."

Damon Sawyer (front) with Dani Wilde and Mike Vernon.

Oli Brown's guitar rig, with mics placed for recording.

» I did offer it to you and you rejected it, so I took it to someone else'. And they said, 'OK, fair enough, but you can't do these two things at once, so you either have to resign or we'll fire you!' So I said, 'Right, I resign as of now,' went away, and about three weeks later I came back and signed an independent production deal with Decca, and that's how I continued on as an independent producer for Decca… and other companies."

The rest, as they say, is history, and Mike Vernon spent the next few years as an independent producer, pioneering one classic blues record after another. After the blues-boom bubble burst at the beginning of the '70s, Vernon started a recording studio in Chipping Norton with his brother, Richard, which continued to be a successful enterprise through to its closure in the late '90s, just a few years prior to Mike's initial retirement.

Back In The Saddle

So now that Mike Vernon has made two fantastic new blues records in the space of the last 12 months, can we presume that this means his self-imposed retirement is effectively over?

"As a result of the Oli Brown record in particular, there have been a few people getting in touch with me and saying, 'We noticed you're back in production, we'd like to send you a demo of this band and that band.' I politely have to say to them, 'No, thank you,' and if they do send them, I politely listen and then say, 'No, thank you!' I'll be 66 in November – and that's young by some peoples' standards, I suppose — but I just don't really fancy going back into it full-time. I would love to do a major production with a major artist. That would be something I would love to do. There's a number of people that I never got to record, and I now won't get to record, because they are getting too old now, BB King of course being one of them. I'd love to be able to make another record with Eric Clapton. I'd love to do that and I think we could probably make a very good record together, but that isn't going to happen either. I don't worry about it any more. I just relax as much as I can and as much as people allow me to, then when something comes along that I fancy doing, and it doesn't appear to be too arduous, I'll give it a shot."

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MIX RESCUE

OUR EXPERTS TRANSFORM YOUR TRACKS

We spend a lot of time rescuing sub-standard recordings and arrangements in Mix Rescue, but this month it really was all about mixing and mastering.

SAM INGLIS

All too frequently, Mix Rescue seems a misnomer for a series that would be more appropriately titled Arrangement SOS, Recording Rescue or Performance ER. In other words, the problems that make a track difficult to mix usually turn out to originate at the pre-production and tracking stages, and require more than tweaking a few faders to sort out.

So it was a welcome surprise to receive the multitracks of Superfox's 'Hear It In My Head', because for once it was hard to find fault with the material, the performance or the recording. The band's catchy blend of pop, soul and funk, which slightly recalls Maroon 5, was clearly well routined, thoughtfully arranged and tightly played. And having hired a professional studio for the basic tracking, they had emerged with a sparkling set of clean WAV files, boasting good tones and excellent separation.

Despite this, the band were having trouble producing a finished product that did justice to the raw ingredients. Eager to get the track done and out to the world, they contacted Mix Rescue to see if we could help.

A quick listen to Superfox's own rough mix suggested a couple of areas for potential improvement. Perhaps most obvious was the low level of the lead vocal, which tended to disappear when the mix got busy. The drum balance also sounded a little odd, and reverb was getting a bit out of hand, so I could see why the band felt the mix as a whole lacked impact and life.

Rescued This Month

Superfox are a four-piece pop-rock band from Sheffield, who have made great strides in the short time they've been together. The full line-up is: Eddie Glossop, lead vocals and guitar; James Glossop, lead guitar and vocals; Harry Soar, drums, and Liam Wragg, bass guitar. A powerful live act, they've played shows alongside Bon Jovi, Little Man Tate, Wishbone Ash and Saul Williams, and their material has been widely aired on local radio.

A Blank Slate

The multitracks happened to arrive just after I'd done a fresh install of Avid's Pro Tools LE 8 on my office Mac. So, armed with an M Box 2 and a pair of headphones, I thought I'd follow Paul White's example from October 2010 and have a stab at mixing the track without using any third-party plug-ins.

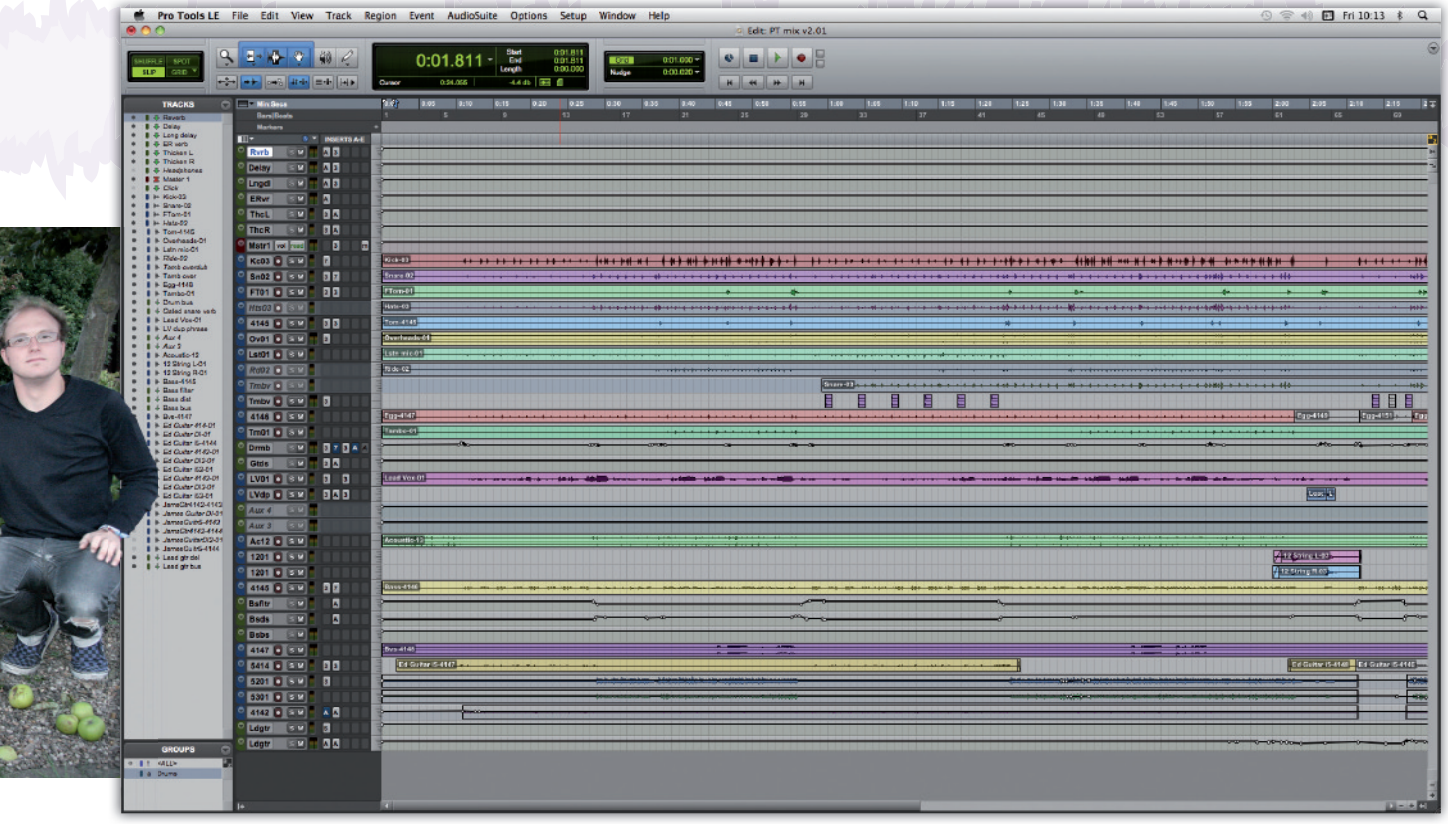
There were 31 audio tracks in total, but it quickly became apparent that not all of these would be needed. For example, the five main electric guitar parts were each represented by three tracks: an amped sound miked separately with an AKG C414 and an Audix i5, and a DI'd version. I decided to reduce each to a single part straight away, so muted the DI and 414 tracks in favour of the i5 versions.

The eight drum tracks included the expected stereo overheads and close mics on kick, snare and (gated) toms, plus three additional tracks labelled 'Hats', 'Ride' and 'Lstn mic'. This last, I assume, was tracked using the listen mic off something like an SSL mixer, and certainly had a character of its own, but I couldn't immediately see a need for the other two, so decided to leave them muted unless such a need emerged later on.

The tracks were all very cleanly recorded, and little EQ seemed necessary beyond high-pass filters on the overheads, toms and snare, plus some mid-range boost to emphasise the attack of the toms. There was, however, no room-mic track, and no perceptible ambience on the overheads, so although it was not hard to get a basic drum balance, it all sounded rather flat and artificial, almost as though it had been built up from samples. Clearly, some reverb was in order, and I ended up using two separate instances of the AIR Reverb plug-in. One was set up as a global effect, generating only early reflections, and I sent a little of the overheads to this to liven them up a bit. The other used a version of the factory 'Drum Room' preset, modified to make it nice and bright. I added sends to this from the snare and from both tom mics, and routed its output to the same auxiliary channel to which the drum tracks themselves were all bussed.


Superfox.

Harry Soar's drumming was very controlled, so compression wasn't required as a corrective measure. However, it's rare to encounter a commercial mix in this genre that doesn't feature compression used for effect, and some quick experiments with the stock Digirack Compressor/Limiter III suggested that a healthy amount of squashing on the drum bus, with a fast attack and release, added excitement and urgency. The snare part in the verses involved a lot of quiet stickwork punctuated by louder hits, and I added a Bomb Factory BF1176 plug-in to its close-miked track in order to balance the levels of the two. I also used the



The entire Pro Tools mix. As you can see from the Track List (left), I've made a number of tracks inactive to render the mix more manageable.

Expander/Gate plug-in to tame the ring of the toms, and to gate the reverb feed so that only the louder snare hits generated reverb. Finally, with lots of guitar tracks to accommodate, I decided to pull in the width of the overheads to make room in the stereo field.

As well as the drum kit tracks, there were two mono percussion tracks: a shakey egg, which was present virtually throughout the song, and a tambourine, which helped to lift the choruses. I hard panned these left and right respectively, and then left them well alone for the time being.

Sing Up

So to the vocals. Listening to Eddie Glossop's lead vocal in solo revealed a rare question mark over the recording itself, in the shape of a noticeable harshness on louder sections. Attempts to create a typical modern rock vocal sound by using masses of EQ and compression simply made it sound scratchy and nasty, so I reined things back to a few dB of gain reduction and a small boost in the 3kHz area, and resorted to extensive level automation to help the vocal sit at the right position in the mix.

Again mindful of the guitar-fest to come, I decided against using reverb on the lead vocal, and instead created two global delay effects — one a short, bright slap-back without much feedback, the other a much longer and warmer panning echo with several repeats. I then used automation to balance the two sends from the vocal track, leaning mainly on the slap delay in the verses and bringing up the longer delay when things needed to be a bit more epic.

The snare and toms had their own, bright reverb, which was gated to stop grace notes triggering it.

I also remembered the old trick of thickening a vocal track by sending it to a stereo pitch-shifter, and shifting the left and right signals up and down a few cents respectively. However, the suite of plug-ins bundled with Pro Tools LE doesn't include a real-time pitch-shifter, so as a substitute, I tried creating two mono aux tracks, panning them hard left and right, and using two instances of the (mono only) AIR Frequency Shifter plug-in to detune them. The resulting effect wasn't the same at all, but in moderation, I quite liked it, and used it on several of the other sources besides.

There was a stereo track of backing vocals, too, which required minimal treatment beyond some reverb and the usual level automation.

Several Six-strings

I didn't anticipate too much trouble from the rhythm guitars, which were all tightly played and well recorded, and so it proved. Most of them appeared only during the chorus, and after a bit of mucking about with faders, they sounded fine to me, apart from one electric part that benefited from a cut at 1.7kHz to stop it fighting with the vocals and snare. Like the drums, the guitars were recorded very dry, so I set »

A lot of automation was needed to make the lead vocal 'sit' against the track. Beneath the main volume parameter you can see where send levels to the delay and 'thickener' buses were adjusted, keeping things much dryer in the verses. The small blue region on the lower track was duplicated from the main vocal track to create a lo-fi delay effect in the breakdown.

» up another global reverb, and also used small amounts of the slap delay and early reflections patches I'd already employed elsewhere. I hard panned all the electric guitars left or right, and that sounded respectable enough, so I didn't attempt anything more subtle.

There were also several lead parts, plus a stereo acoustic guitar, which opened the track and reappeared during the choruses. The acoustic sounded great just as it was, but needed narrowing to make sense within the context of the mix. In the verses it was joined by two lead parts, which again I panned hard left and right and anointed with slap delay and early reflections. One of these was a busy, clean funk part; a little compression with a shortish (4ms) attack time helped the transients pop out, while I used level automation to ensure that the details could be heard and to duck some of the 'chuck-a-chuck' damped sections between phrases. The other was an almost keyboard-esque melodic part, which, to my ears, cried out for some sort of 'ear candy' treatment to make it more engaging. I found this in the AIR Vintage Filter plug-in, using its envelope follower to set up something a little like an auto-wah in the verses. I bypassed this in the chorus, there being plenty else going on there to engage the listener's attention.

Finally, there was a proper overdriven lead guitar, which executed a nice solo after the middle eight and some melodic twiddles over the final chorus. Guitar solos are often an opportunity to wheel out some fun effects, and this was no exception. I panned the source track hard right, then sent it to a mono aux panned hard left; this had a short delay set 100 percent wet, taking advantage of the Haas effect to create a wider sound. Both of these tracks then fed a stereo aux channel adorned with an AIR Dynamic Delay plug-in, set up to create a long ducking delay, and a flanger to add some movement. Plenty of level automation was also required to get the guitar solo to sit nicely with the vocals and the rest of the mix.

Split Personalities

The last major instrument to worry about was Liam Wragg's bass guitar. As is often the case, this took some time to get right, even though it was well played and well recorded. There was quite a contrast between the bass part Liam played in the verse, which was punchy and staccato, and his more legato chorus bass line, and any mix treatment that worked well for one didn't seem to work for the other. In the end, I created two different aux channels with different effects on them. One used the AIR Vintage Filter to add low-mid thickness and substance, while the other employed the AIR Fuzz Wah plug-in to add grit and bite. I then used automation to balance these and the lightly compressed dry signal to taste: the filtered version helped add substance in the verses, while the distorted version helped the bass cut through the wall of guitars in the choruses.

Drop Dead

With the basic mix done, I began to feel that there was one area where the arrangement could use a little help. After the middle eight and guitar solo, there was a drop, where the band returned to the sparse feel of the opening verse before kicking back into the final chorus. It seemed to me that the effectiveness of this drop could be enhanced by a little bit of creative editing and processing. I cut out and muted the relevant sections of the shakey egg and funk guitar tracks, and used another instance of the AIR Vintage Filter plug-in to screw down the drum bus, the aim being to achieve maximum impact when the big drum fill kicked back in at the end of this section. To emphasise the spacey quality of the drop, I also duplicated the lead vocal line 'To know that we'll always be friends', moved it to a new track, shifted it a bar later and applied the AIR Lo-Fi plug-in to create a ghostly one-off delay effect. An hour or so of fiddling with plug-in »

In the absence of a conventional pitch-shifter, two instances of the AIR Frequency Shifter were used to create a stereo 'thickening' effect.

Audio Examples Online

We've placed some 'before and after' audio files on the *SOS* web site so you can hear for yourself the changes that Sam made to Superfox's track. Go to www.soundonsound.com/sos/dec10/articles/mixrescueaudio.htm to hear them.

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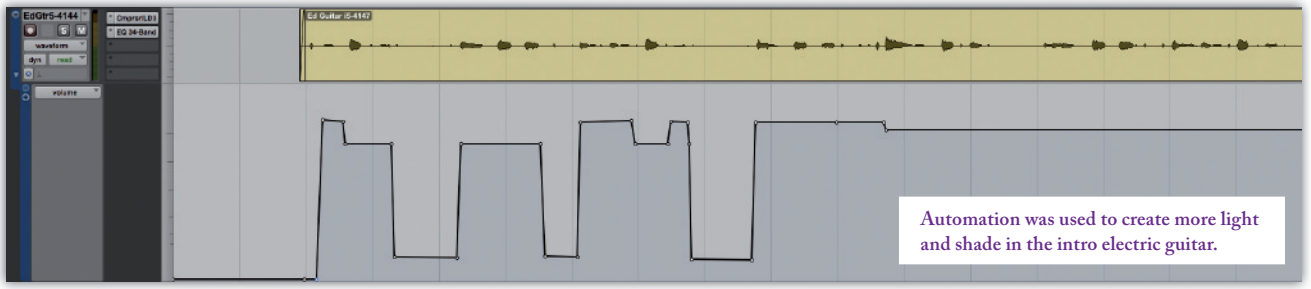


Automation was used to create more light and shade in the intro electric guitar.

» automation later, and I felt I had something that really added to the track.

The Final Touches

The final stage was to apply some sort of master processing to bring things up to a respectable level, and hopefully to add some beef. Up to this point, I had felt more liberated than limited by the lack of third-party plug-ins on my computer. The AIR effects sounded good, and the Pro Tools EQ did its job well. However, while the Compressor/Limiter III plug-in is easy to set up and gave clean, transparent results on individual channels, it can't compete with some of the more luxurious third-party mix compressors on the market, and the only dedicated mastering plug-in bundled with Pro Tools LE is the ancient Maxim limiter. I used these to bounce out a rough mix to play to the band, but decided that final mastering would have to wait until I had access to something a bit better.

Remix Reaction

Superfox's Harry Soar comments: "When we tried to mix our track, we couldn't find a way to get the balanced, powerful sound that we as a band love. Radio-friendly bands seem to find bags of depth and character in their mixes. Yet we couldn't find such depth in our track and we couldn't put our finger on what it lacked.

"Then we heard the new mix and we loved it immediately! It managed to balance all the many different sounds in the mix without losing sight of the song. The extra echoed vocal lines and reverb effects made the sound much more rounded and polished, and the snare effect in the middle eight was a great touch. These little things helped to create a mix that has more "flavour". The direction Sam has taken was spot on, and exactly what we were hearing in our heads (no pun intended...) when we thought of the track. We can't wait to get people listening to it, so it'll be going up on Spotify, iTunes and our MySpace page very soon!

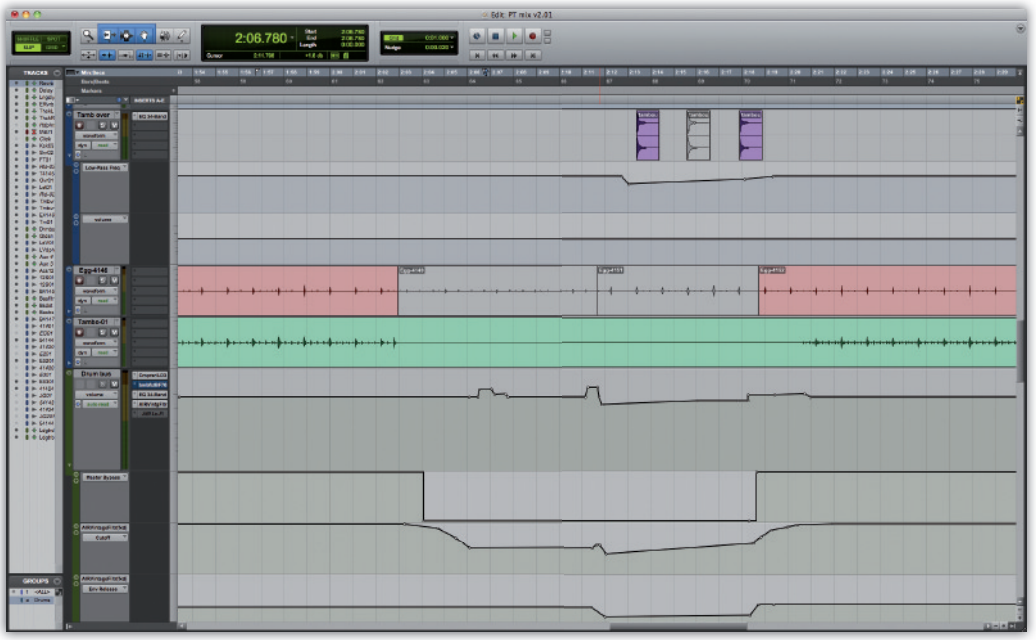
"Thanks so much to Sam at *Sound On Sound* for taking the time to work some magic on our track. You're a legend!"

Happily, the band liked the rough mix, and had only a few changes to suggest. They had the excellent idea of adding a tambourine sample to reinforce some of the snare hits in the verses, which worked well; they also spotted that the last note of my ghostly vocal delay in the drop clashed with the underlying harmony, so I used the off-line AudioSuite pitch-shifter to push it down a semitone.

Until this point, I'd only heard the mix on headphones, so I took a raw 24-bit master home with me, partly so I could check it on speakers and partly to take advantage of some proper mastering tools. I'm not usually a fan of multi-band compression, but in this case, Waves C4 did a good job of thickening things up without affecting the essential liveliness of the sound. I also applied a touch of the same company's S1 stereo width enhancer, and then ran the mix through the tape emulation section of Magix's AM-Track, which is quite a good way of making a mix louder without too many adverse side-effects. For the final limiting process, I turned to the new Slate FGX plug-in, which is gobsmackingly effective at making

A ducked delay and a flanger helped to add size and movement to the lead guitar.

Two different auxiliary channels were created for processing the bass guitar. One (left) used a filter to help thicken up the low end, mainly in the verses, while the other applied distortion to help it cut through in the choruses.



The only real arrangement tweak I made was to emphasise the breakdown after the middle section. The egg and tambourine tracks were muted (top) and automated effects were applied to the drum bus. You can also see the sampled tambourine hits that were added at the band's suggestion.

things loud while retaining the integrity of transients and other dynamic elements.

If It Ain't Broke

In the end, then, this month's project turned out to be less about rescuing and more about just mixing. As such, I think it underlines how crucial it is to get things right in the first place. When a band who can play use a good studio to record quality material, as was the case here, it becomes so much easier to steer the mix back onto the right path, even without the aid of fancy third-party plug-ins. Mixing a well-recorded track is primarily a matter of balancing levels, with processing and effects used to enhance good source material rather than disguise its faults. If only it was always like this!

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Mode Machines KRP1 Krautrock Phaser

Analogue Phaser

The KRP1 Krautrock Phaser is a modern recreation of a German classic.

GORDON REID

Many of the sound effects that we now take for granted actually appeared as recently as the 1970s. The chorus ensemble was the first of these, and was soon followed by the phaser; most notably by the Roland Phase 5 and MXR Phase 100 in 1974, and the Electro-Harmonix Small Stone the following year. Nonetheless, there appear to have been a handful of earlier, more experimental phasers, and the country in which their sonic boundaries were most likely to be pushed to the limit was Germany, the home of electronic music. German bands were often more interested in processing sounds than they were in playing dozens of notes in the shortest possible time, and pre-eminent among these was Tangerine Dream, whose trademark 'whoosh' was created by playing a Mellotron through a hand-built German phaser called the Gert Schulte Audio Elektronik Compact Phasing 'A'. You can't overestimate the importance of this unit, and some people have suggested that it was as important as the band's keyboards and synthesizers. Shortly thereafter, Kraftwerk adopted the 'A', while in the plank-spanking

»

Mode Machines KRP1 Krautrock Phaser £399

PROS

- It has a character and sonic quality that distinguishes it from other phasers.
- It generates a surprisingly wide range of effects, from deep sweeps to subtle 'sparkle'.
- It's chunky, heavy and feels very robust.

CONS

- Its operation is not always clear, and the documentation is not always explicit.
- Arghh! Yet another external power supply.
- It's expensive.

SUMMARY

There are lots of cheap, not very nice phasers on the market, plus a tiny handful that are nicer and more expensive, but very few set your ears a-tingling. Although the KRP1 is possibly the most costly of them all, it can sound superb and it has bags of character. If you're looking for effects with a bit more polish than the norm, you have to try it.

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» world it was used by none other than Richie Blackmore of Deep Purple, so it was surely only a matter of time before a clone would emerge. It has taken nearly four decades for it to do so but, finally, here it is: the KRP1 'Krautrock' Phaser.

Out Of The Box...

The first thing I noticed about the KRP1 is that it's heavy. Despite being hand-made (which I think is sometimes a mixed blessing), it feels robust, and while one would have to be brave or foolish to take an original Compact Phasing 'A' on the road today, I would have no hesitation in bolting a KRP1 to a pedalboard and shoving it in the back of a van. Unfortunately, I was less happy when I found its wall-wart, because I had expected an integrated power supply. Another thing I should point out is that the wooden end-cheeks shown in the photographs are optional extras and cost an additional £20.

Setting up the KRP1 was, of course, trivial. Plug in a mono source, one or both outputs, plus the DC supply, and you're ready to go. Happily, the input is not an exact clone of the A's, because it has been tweaked to accept a wider range of signal levels, resulting in cleaner and brighter results from low-level instruments such as guitars and Pianets.

As you can see, there are just four knobs. The first three of these are easily understood; they control the width of the sweep, the period of the sweep, and the depth of the notches, while the large lamp on the panel gives you an immediate indication of the sweep rate and your position within it. However, obtaining a specific effect is less than straightforward, in large part because even the manufacturer appears to be unsure what happens when you adjust the fourth knob, labelled Modulation. Some people, including the presenter of Mode Machines' own on-line demo, have described this as a Mix knob, controlling the relative amplitudes of the dry and wet signals in stereo use, but simple tests demonstrate that this is not the case. It sounds as if it is applying a slight phase shift or detune between the channels, although the effects that you can obtain from it — which include mild pseudo-stereo panning — can depend upon the positions of the other controls.

On the back of the unit, you'll find two control inputs and a switch. In principle, their operations are self-evident. The first allows you to control the amount of phasing using a foot (volume) pedal, while the second allows you to control the rate using a second. The switch, labelled Auto/Manual, suggests that you can bypass the phasing depth and rate controls and treat the KRP1 as a sophisticated wah pedal, but that's not quite correct, because there are »

Alternatives

In principle, any phaser is an alternative to the KRP1; a vintage phaser even more so. But its most significant competition lies in the form of the thoroughly contemporary **Moogerfooger MF103** 12-stage phaser. This is a much more flexible device, since it offers two phasing modes (six-stage for three notches, and 12-stage for six notches), a wider range of sweep rates, overdrive, level matching between the input and affected signals, and CV control over all four main parameters: Rate, Amount, Sweep and Resonance. So why would anyone look at the simpler and less flexible KRP1? The answer is, of course, because they sound different.

The History Of The KRP1

The KRP1 is the brainchild of a German engineer, Jurgen Haible, whose recreations of interesting analogue processors and synthesizers are highly respected among synth enthusiasts.

He first cloned the Compact Phasing A in 1999. The development had been financed by MAM (Music & More), whose co-owner Michael Thorpe had approached him to discuss the possibility of manufacturing the finished unit as a MAM product. But MAM were in financial difficulties, and although Thorpe demonstrated the KRP1 prototype at the NAMM show in 2000, the project was shelved.

In 2007, Haible began selling the clone as a DIY kit, but in 2009 he and Thorpe decided that there was room in the market for both the kit and the finished product. Unable to decide whether to reinvent MAM or use Thorpe's TBS (Touched By Sound) brand, they chose instead to create a new brand, Mode Machines, which now carries the KRP1, the Xoxbox (apparently pronounced 'socks box') clone of the Roland TB303, and the SL1 Synthlab analogue monosynth.

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» still interactions between the top panel knobs and any values determined by the pedals. By the way, the fledgling documentation did not mention this, but I found that I needed to use pedals with TRS jack plugs for everything to work. When did phasing become so complex?

The Sound

Given its background, I thought that it would be interesting to compare the KRP1 with two vintage phasers, an original Small Stone and a Roland Phase Five. Despite the fact that all these units operate on the same principles, the sonic differences between them are immense, although, when you consider the unusual architecture of the KRP1, this maybe isn't as surprising as it first appears.

Experimenting with the KRP1's stereo output channels (the EH and the Roland are strictly monophonic), I patched it through a mixer to two sets of speakers: a wide, stereo pair of active monitors, and a stage combo with stereo inputs but just a single speaker enclosure. I found that the sound of the Left channel was often — although not always — different from that of the Right, which in turn could be different from L+R played in stereo, which was again different from L+R mixed to mono. While these differences were often subtle, sometimes they were not, and I found that they greatly extended the range of effects available.

In use, it soon became apparent that the KRP1 has a wider frequency response than either of the other phasers, resulting in a cleaner, brighter sound. Harder to describe, the KRP1 also has a distinctive character that is quite unlike the EH and the Roland. This is particularly apparent when applying deep sweeps to harmonically rich sounds, and you can hear why musicians in the 1970s used the Compact Phasing A with their Mellotrons' strings and vocal tapes, as well as with the string synths of the era. What's more, the KRP1 is capable of producing the 'clipped' sweep of the Compact Phasing A, where the notches are so high up the spectrum at the top of the sweep that the phasing effect appears to pause at the top, then rush down, sweep up, pause and continue again. It's very distinctive, and Tangerine Dream fans will recognise it instantly. I suspect that, when Tangerine Dream first passed their Mellotron's '3 Violins' through an 'A', it would have been almost impossible to avoid composing *Phaedra*.

The KRP1's other great strength lies in the range of subtle effects that conjure sparkle and *je ne sais quoi* for instruments such as guitars, Clavinets, Rhodes pianos and Pianets. Used sympathetically, it adds fairy dust with a delicacy that other phasers cannot.

Inevitably, not all is hunky-dory, and I found that you have to be careful when using the KRP1 because, while it looks like a new piece of kit, it retains many of the foibles of the original. As well as the oddities of the Modulation knob and the foot pedals, the feedback loop is not constrained to gains below which self-oscillation occurs, so beware ear-shredding screams if you're not circumspect in its use. You also have to be careful to avoid distortion with some signals and, although the unit is not overly hissy, it was prone to generating hum when connected directly to my active monitors. What's more, the affected signal was often significantly louder than the input, and whereas the original Compact Phasing A could fade its effect in (or out) when you switched it on (or off), the KRP1 does not; the stomp button is a hard bypass so you get a significant discontinuity when you jump from the unaffected to the affected sound (or vice-versa). Finally, the external PSU started to buzz after a few hours operation. I hope that this is just a one-off fault with the supplied unit, but it's worth noting nonetheless.

The KRP1's rear panel, featuring footpedal inputs for amount and rate, two quarter-inch outputs, a single quarter-inch input and a socket for the external PSU.

Secret Knob?

I asked Mode Machines' Michael Thorpe what the Modulation knob does, and he told me, "We came to the conclusion that it does not have a fixed function. It seems to work in combination with the knobs next to it, and its effect also seems to depend on what frequencies are present in the input signal. We discovered this problem years ago when we first thought about cloning the Compact Phasing A, because we couldn't tell what the function was. But comparing the KRP1 to the original units, we are sure that we have cloned it correctly. I think that Gerd Schulte didn't know what he was trying to make in the early days! Perhaps we had better call it the 'Secret Knob' from now on."

Conclusions

For some reason, I had approached the Krautrock phaser with a certain degree of disinterest. I was also sceptical that, in a world replete with usable phasers costing less than £100, a clone of a long-defunct and little-known unit could justify anything like a price of £399. Then it took me much longer to get to grips with it than I had expected.

So here goes... I admit it, I was wrong. While not perfect, the KRP1 sounds great and — whether you're using it to create deep, whooshing sweeps or add delicate sparkle to signals — it works beautifully with both guitars and keyboards. It's not cheap, and there are other high-quality phasers available, but this is one of those rare cases where the idiosyncrasies of the product add to its character rather than detract from its usability. It's not for everyone, admittedly, but if you're in a Tangerine Dream tribute band, you have to buy one of these. If you're not, you're still going to be tempted.

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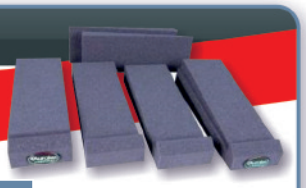

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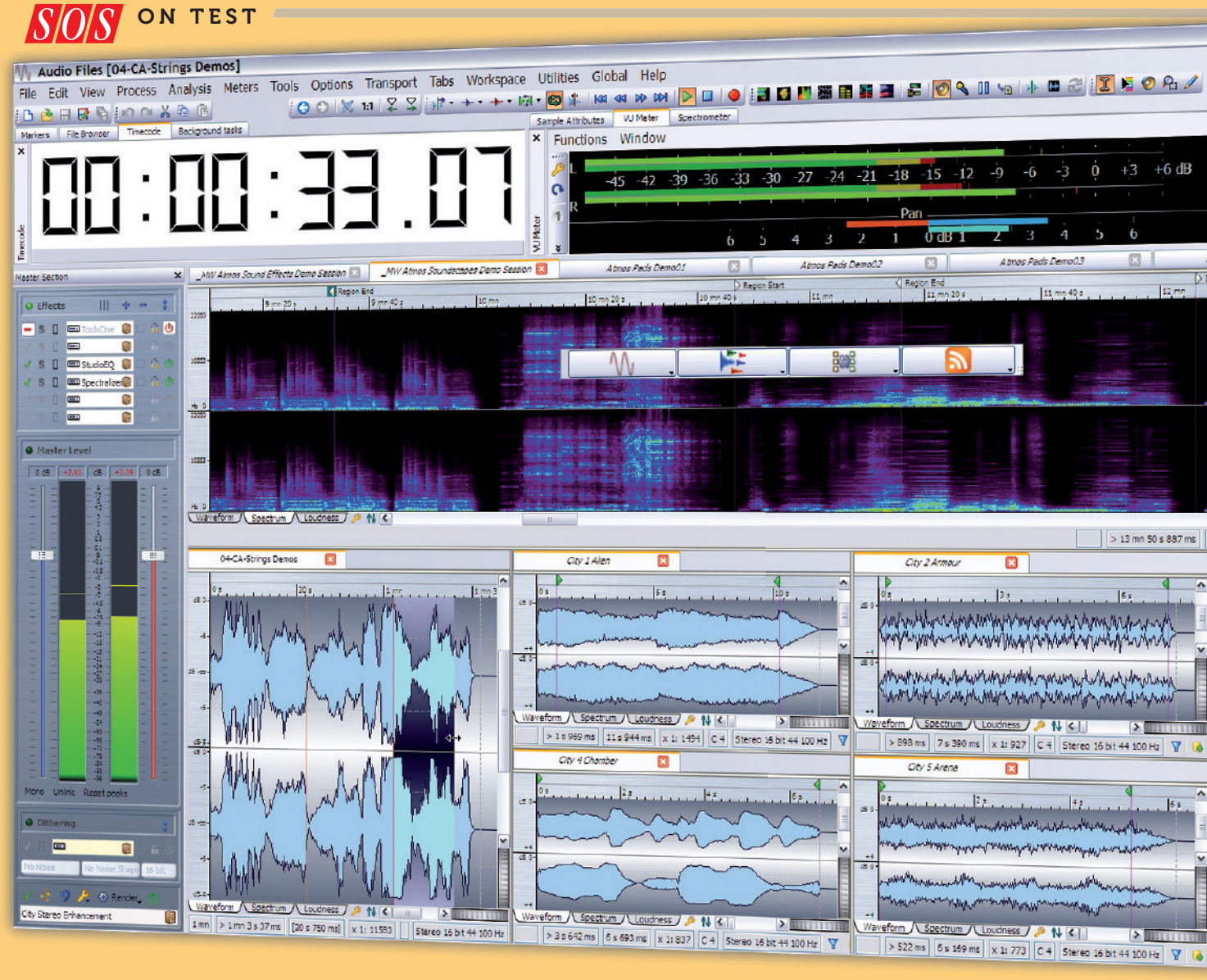

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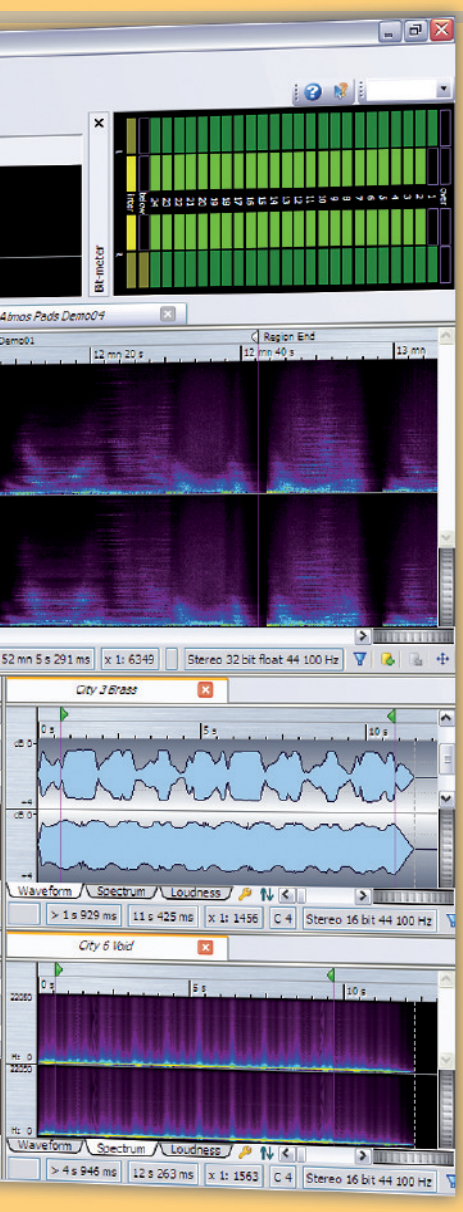
I've been using Steinberg's Wavelab for audio editing, mastering, and restoration on an almost daily basis since version 1.6, way back in 1997 when CD burning and real-time plug-ins were first added, and have watched it grow from a good stereo editor, through the excellent sampling, looping and analysis functions of Wavelab 2, the multitrack Montage additions of Wavelab 3 for assembling layered files and compiling albums from individual tracks, the graphic redesign and plug-in bundle of Wavelab 4 and the DVD-Audio and video montage support of Wavelab 5, to the eye-boggling spectrum display and editing options of Wavelab 6.

It's still largely the work of one man, Philippe Goutier, and each time I review a new version I wonder how on earth he can improve it any further, but with Wavelab 7 he's done it again, and in a big way. For the first time ever, Wavelab is now cross-platform, enabling Mac users to directly experience what all the fuss has been about for so many years, but there are a lot of new features for PC users as well, from a new workflow concept, high-end restoration tools and 30 VST3 format plug-ins to a state-of-the-art CD-burning engine.

Seventh Heaven?

Wavelab 7 has a different look to its predecessors, with a rather more subdued, yet sophisticated graphic feel. I found



Wavelab 7 sports a new and more sophisticated graphic look that's less distracting on the eye, a neat new set of tabbed Tool windows, and flexible Tab Group options to arrange your files on screen. Notice also the four buttons of the Windows Switcher, here superimposed on the spectrum editor display towards the centre of this screenshot.

this really helped me focus on the job in hand, compared with the *faux* metalwork of Wavelab 6. Some long-term Wavelab users might find the new look and layout disorientating initially, but it didn't take me long to adjust, and having done so I would never go back to the now clunky-looking Wavelab 6. There are also many graphically enhanced icons to take in on the customisable toolbars, and lots of new customising options to overwhelm the newcomer, although once you get to grips with them all you'll wonder how you ever managed without them!

The most obvious addition is the Windows Switcher, a small and resizeable floating window that remains visible at all times — even, by default, after switching to another application, although you can disable this if you wish — and which lets you leap straight into audio file editing, multitrack montage, batch-processing or podcasting duties. For instance, even while writing this review I have the Wavelab 7 Switcher superimposed over the title bar of my word processor, and can create a new file, open an existing one, or simply open an empty workspace in any of these categories. I took to this like a duck to water.

While you could launch multiple instances of Wavelab 6 to simultaneously work across several projects, this ability has also been seriously upgraded in Wavelab 7, since each new workspace you open is a separate entity, enabling you to quickly switch between audio editing, album compilation, batch processing and podcast creation at will. There are also several dozen editing, viewing and analysis functions that you can now open in the new Tab area inside tabbed 'tool windows', covering such things as various metering alternatives, marker creation and editing, file browsing, error detection and correction, spectrum editing options and so on. These tool windows provide great versatility: you can open and close them in stand-alone 'floating' mode above your workspace, as in previous Wavelab versions, or drag them to various on-screen locations and dock them. »

Steinberg Wavelab 7
£497

PROS

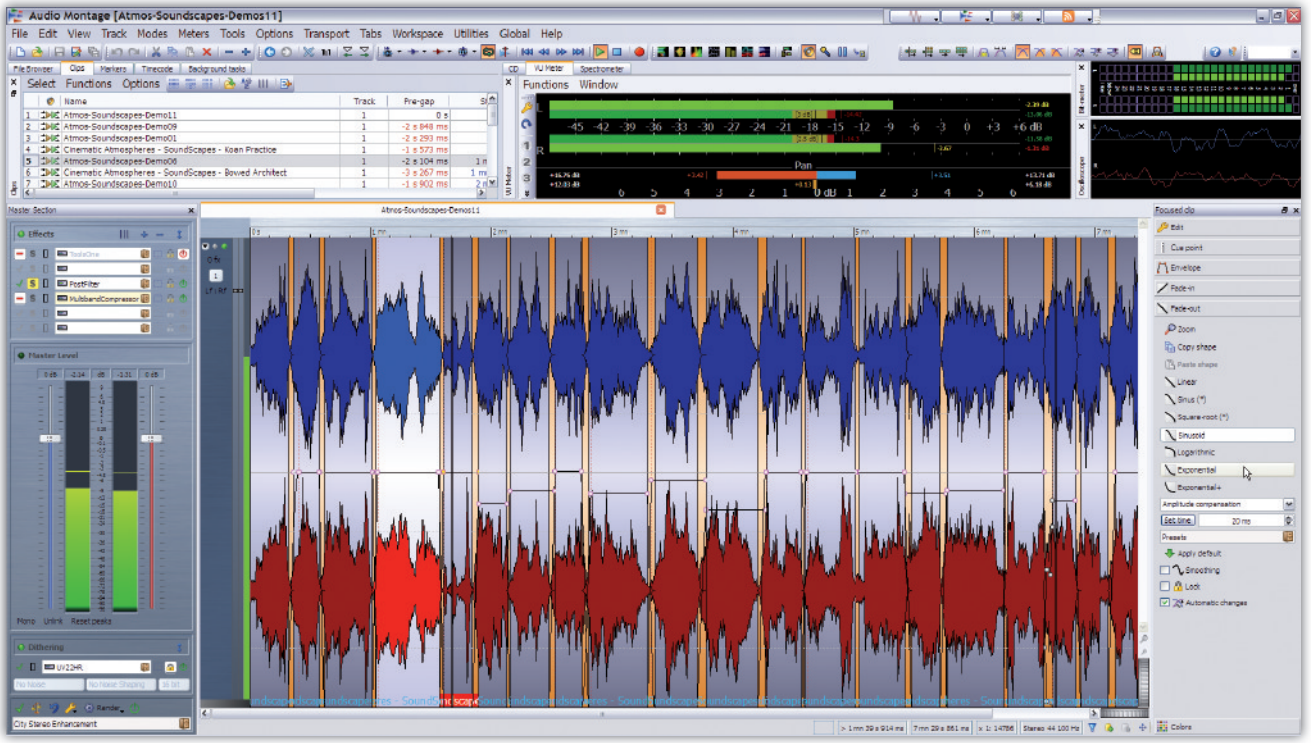
- Four extremely versatile new workspace environments.
- Very flexible resizeable and tabbable window arrangements.
- Sonnox Restoration plug-in suite.
- Much improved multi-core engine.
- Now available for Mac!

CONS

- No official support for Windows XP or Vista.
- No longer any printed paper manual.
- Still no plug-in automation.

SUMMARY

This is a stunning update to an already mature product that should win plenty of new converts, especially now that it's finally available for Mac as well as PC.



With 'Focused clip' menus on the right, a much more configurable Master Section on the left, and numerous tabbed Tool windows available across the top, Montages are even easier to assemble than before.

The latter approach is a great improvement, since you can carefully arrange your most-used tool windows in tabbed sets (see screenshots), using some natty animated options to drag and drop the various tool-window sections where you wish, while all the others move themselves out of the way to make room. Horizontal and vertical zooming functions now have thumbwheel graphics for easier control, and respond to your mouse scroll wheel when the cursor is superimposed on these graphics.

Making Arrangements

With so many people now working with larger screens, the rather basic Wavelab 6 'cascade', 'tile horizontally' or 'tile vertically' options to arrange the various open windows have now been seriously upgraded by the new 'Tab Group shortcuts', which offer various on-screen arrangements of up to nine windows, each supporting multiple tabs, so you can easily work with dozens of simultaneous audio files grouped into different batches. You can also create your own Tab Group arrangements by splitting an existing group, horizontally or vertically, using the icons in each workspace area.

If you regularly edit audio alongside other applications, the new 'Position on screen' option also helps to streamline your working methods by offering 25 predefined choices for each instance of the Wavelab 7 application, ranging from the 'Full screen view', to (for example) Wavelab occupying the top or left half of the screen, or the bottom right-hand quarter of the screen. This flexibility makes it very easy to create split-screen views with your sequencer application, Internet browser, or even tiling multiple instances of Wavelab 7 itself.

Better still, your multiple window arrangements can now be saved as workspaces, so you can optimise everything to suit different aspects of your work, such as sound recording, editing, restoration and mastering, with everything resized, repositioned and tabbed to perfection. This is particularly helpful to those using multiple monitor screens, but even those with a single monitor will find they make better use of the available space when a selection of carefully tweaked layouts is only a couple of mouse clicks away.

The Montage is also much more streamlined now, partly because of the new Workspace flexibility, but mainly because of the new 'Focused clip' menu: right-click anywhere over a montage clip and this new menu appears, displaying collapsible 'accordion' panes covering Edit, Cue Point, Envelope, Fade-in, Fade-out and Colour options. I found these much quicker to use than Wavelab 6's floating nested menus, since you can always see all relevant parameters. Even better, most of these options now have descriptive graphic icons as well as text descriptions; in the case of the fade-in/out settings, for instance, this makes choosing the most

System Requirements

Wavelab's minimum requirements are comparatively modest. The box states that it only needs 200MB of hard drive space, and runs on Windows 7 with 2GHz CPU and 1GB RAM, or Mac OS 10.6 'Snow Leopard' with an Intel CPU and 1GB RAM. However, practical day-to-day requirements for RAM and CPU may rise considerably if you want to load in lots of simultaneous plug-ins, and Philippe Goutier himself suggests a minimum of a dual-core CPU and 2GB RAM, while actually recommending a quad-core CPU and 4GB of RAM.

One important thing for PC users to note is that only Windows 7 is officially supported — if you're running Vista or XP you're on your own support-wise, although in practice I experienced absolutely no problems running Wavelab 7 under Windows XP. Existing users of Wavelab 6 can run the two versions side by side, and Wavelab 7 can also import Master Section presets created in its predecessor, but, sadly, no other presets.

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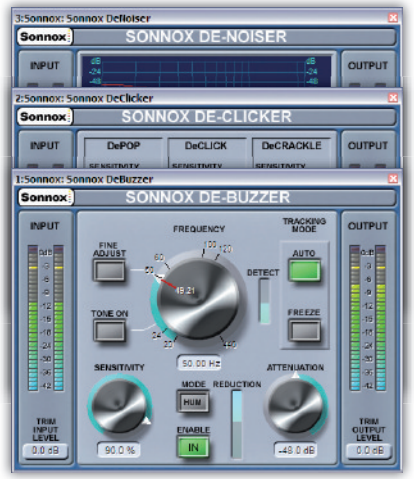
Master Of The Universe

The Master Section, with its multiple effect plug-in slots, master faders, metering and dithering sections, also sports some handy new additions. These include dedicated buttons to toggle individual plug-in windows between visible and invisible status, locking of individual plug-in slots so they ignore the bypass function and when loading new Master Section presets, and the ability to force the Master Section meters to display the output level at any stage of your plug-in chain (useful for detecting overloads and so on). Like the other tool windows, the Master Section is now resizeable, so you can finally adjust its width to display plug-in names in full, or make the master faders and meters as tall as you like.

The smart bypass options that match in/out levels in various ways, so you can A/B with or without effects at the same subjective level, are now part of the Master Section itself, and this function cleverly defaults to the 'A' and 'B' keyboard shortcuts, so you can perform A/B tests without a second thought. Wavelab's Edit menu now also includes the option to store a set of Master Section plug-in presets alongside an audio file, which is handy if you don't want to permanently render the treated audio: when you reload that audio file, all your plug-ins and their settings will reappear ready for further use.

Overall, I found this new Master Section a lot more flexible, although a few controls weren't quite as clear as in Wavelab 6, such as the old Bypass button, which has been replaced by a rather anonymous green tick that merges into the background, and the mono button, which is now a simple text label. The general approach in Wavelab 7 seems to be to have visually subtle default settings, while making any non-default settings more obvious, which does make complete sense once you get used to it.



The biggest news in the plug-in department is the arrival of a suite of three restoration tools from Sonnox.

Some keyboard shortcuts have also changed to more obvious settings, although if, like me, you're too used to the previous shortcuts, it's easy enough to change them back in the 'Customize commands' area of the Options menu. Don't be too hasty, though, as once I got used to the new ones I did find most more logical and easy to remember.

Processing & Analysis

The Dirac processing options offered by Wavelab 6 for time-stretching, pitch-correction and pitch quantising were already some of the best around, providing a suite of settings from 'time localisation' to suit solo instruments and drum loops, to 'frequency localisation' for more complex mixes and classical music. I've used them extensively, even on sensitive material such as choir sample libraries, but Wavelab 7 beefs things up still further. Its previous generic lower-quality options have now been abandoned in favour of the latest Dirac 2.2 algorithms, which offer various »

Previous Wavelab Reviews In *SOS*

Wavelab 6: June 2006
W www.soundonsound.com/sos/jun06/articles/wavelab6.htm

Wavelab 5: February 2005
W www.soundonsound.com/sos/feb05/articles/steinbergwavelab5.htm

Wavelab 4: May 2002
W www.soundonsound.com/sos/may02/articles/wavelab4.asp

Wavelab 3: March 2000
W www.soundonsound.com/sos/mar00/articles/wavelab.htm

Wavelab 2: June 1998
W www.soundonsound.com/sos/jun98/articles/wavelab.html

Wavelab 1.6: October 1997
W www.soundonsound.com/sos/1997_articles/oct97/steinbergwavelab.html

Wavelab 1.5: February 1997
W www.soundonsound.com/sos/1997_articles/feb97/steinbergwavelab.html

Wavelab 1.01: August 1996
W www.soundonsound.com/sos/1996_articles/aug96/steinbergwavelab.html

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Preamps . The UFX provides a battery of four high-transparency preamps backed up by four stereo converters with RME´s new Advanced Parallel Converter technology. The fully symmetrical preamp design impresses with extremely low distortion, excellent signal to noise ratio and a perfect linear frequency response. The four inputs on the front use balanced XLR/TRS combo sockets and operate alternatively as Hi-Z inputs. Every channel uses a parallel AD conversion on two AD converters at the same time to reach exceptional signal-to-noise values and a performance not available with a classic converter design.

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TotalMix FX . The DSP-based TotalMix mixer allows fully independent routing and mixing of all 30 input and playback channels to all 30 physical outputs. The mixer not only matches features of high-end digital consoles but even adds hardware calculated effects like 3-band parametric Equalizer, Dynamics, Auto Level plus Reverb and Echo to the mix. All effects are available even at 192 kHz.

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Test Spec

- Steinberg Wavelab v7.0.
- PC with Intel Conroe E6600 2.4GHz dual-core processor, Intel DP965LT motherboard with Intel P965 chip set running 1066MHz system bus, 2GB Corsair PC2-6400 DDR2 RAM, running Windows XP SP3.

» code enhancements, as well as additional options to modulate time-stretching and pitch formant-correction over time in a graphic window, and preserve formants when pitch quantising. For those working in more extreme audio editing environments — such as sound designers, who may routinely pitch-shift by several octaves to create special effects — these tools now also support sample rates up to a massive 384kHz, as does the Crystal Resampler tool.

The intelligent Loop Tweaker has been enhanced with a Stereo Merge option, so you can view both channel loop points superimposed, while 'Display processed audio' lets you visually toggle your before/after tweaks in the Tweaker window to see how much better your looped waveforms 'join up'. I'd still like to see larger zoom options so you can zero in on the join even more, but this remains a very accomplished tool for sample-library developers. Meanwhile, although the already excellent Global Analysis tool for measuring peak and RMS levels and spotting basic errors is largely unchanged, the Error Correction tools have now moved into the tool window selection options, while the 3D Frequency Analysis display can now be rotated by dragging a thumbwheel, to help you more easily spot audible anomalies.

New Plug-in Goodies

The majority of the 30 new bundled VST3 plug-ins will already be familiar to Nuendo users, and many to Cubase users too. For other sequencer users, suffice it to say that they include a very effective selection of EQs, single and multi-band compressors, chorus/delays, and special effects such as an Envelope Shaper and Octaver, plus tools to (for instance) mix down surround mixes to stereo. Of special note are Nuendo's very flexible Post Filter, with its low/high cut and up to eight notch filters for cleaning up audio material, the four-band Multi-band Compressor and four-band parametric Studio EQ, and the versatile Roomworks reverb.

This was also my first experience of the Sonnox restoration plug-ins reviewed in *SOS* March 2010, and although the De-Noiser, De-Clicker and De-Buzzer on offer here are seriously cut-down compared with the full £1400 Sonnox Restore suite, I nevertheless found them extremely effective and worthwhile additions. In particular, the De-Noiser, while not the most effective in my personal collection (that accolade still goes to Wave Arts' MR Noise), is a huge improvement over the Spectral Design version offered with previous Wavelab versions. The only thing I still miss in Wavelab 7 is automation for plug-in parameters: if you want to morph any effects across your audio clips, you'll still have to export them to another application to do so.

Format Options

Thankfully, for PC users, Wavelab 7 retains support for DirectX plug-ins in its Audio workspace (though still not in Montages). These may be long in the tooth now, but some of us still have must-use items that are unavailable in VST format, and it's good to see this continued support, especially considering that DirectX has been abandoned in Steinberg's Cubase. The Apple OS X version of Wavelab can also natively open AAC, M4A and M4P-format audio files using its proprietary codecs, and PC owners can do the same if they install Apple's free QuickTime (normally installed with iTunes on a PC).

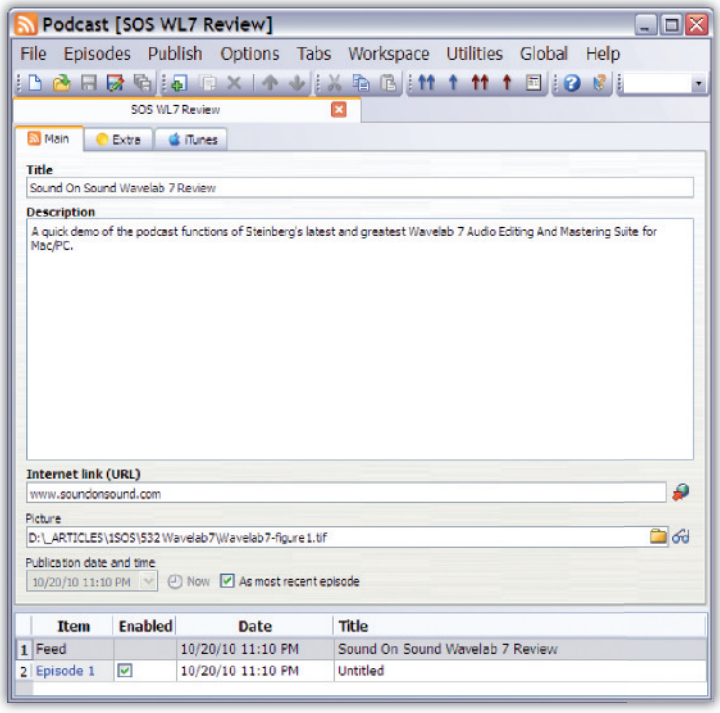
Podcasting was already available to Wavelab 6 users, and the commented 'episodes' in Wavelab 7 offer similar compression options to ensure smooth delivery of streamed audio on-line, but with the addition of some useful extras, such as suitable extensions to make your podcast compatible with iTunes and its categories, complete with author name and contact details. You can also publish your podcasts directly from Wavelab 7 by entering your FTP site upload details.

Meanwhile, for those wanting to output directly to a CD-R, Enhanced CD or DVD-Audio disc, Wavelab 7 features a completely rewritten burning engine designed for greater reliability, which for the first time allows you to burn an audio CD from an industry-standard DDP (Disc Description Protocol) image file, as well as offering DDP as an output format, for reliable error-protected transfer of files intended for optical disc duplication that can even be safely transmitted across an Internet link.

Moreover, Wavelab 7 supports the MPEG 1 Layer 2 (Musicam) file format, which is commonly used in digital broadcast systems, as

Alternatives

On the PC, the most obvious competitor for Wavelab is still **Sony's Sound Forge** (now at version 10), which has (thankfully) reverted to including the excellent CD Architect Red Book CD-mastering functions in its arsenal, or you could look at all-in-one applications like **Magix Samplitude** (or its bigger brother **Sequoia**), and dedicated and rather more expensive systems like **Sonic Solutions** and **SADiE**. On the Mac, the main competition is probably **BIAS Peak**, arguably the most comprehensive stereo editor available on this platform, and, like most other packages mentioned here, available in various versions. Each of these applications has its own devout following, but Wavelab remains a very strong contender.



The podcasting options now include extensions to make your podcast compatible with iTunes, and you can also publish your podcasts directly from Wavelab 7 by entering your FTP site upload details.

well as Broadcast WAV, the extended WAV format that includes additional information in the file header. Steinberg really do want as many professionals as possible to be able to take advantage of this latest cross-platform application.

Multi-core Improvements

Wavelab 6 could only utilise a single processor core for all the plug-in effects, while a second core was used for the graphics and other functions, and on machines with four or more cores you could quickly hit a CPU ceiling, even though you still had plenty of processing power in reserve across the other cores. Thankfully Wavelab 7, both on Mac and PC, now makes far more intelligent use of all available cores. For instance, my PC could only run a single instance of 2C Audio's excellent Aether reverb in Wavelab 6 without glitching, even though my CPU had plenty of spare power in its other cores. In Wavelab 7, it easily manages five instances before anything untoward occurs.

Batch processing is an indispensable suite of Wavelab functions for sound designers and editors, letting you apply the same series of editing or conversion processes to folders full of audio files and then save them one by one in your chosen format. The improved multi-processing engine now runs its batch activities as a background task, so you can carry on editing while your collection of files is processed. You can even specify how many of your cores are allocated to batch processing, to get the desired balance between time taken and user interface responsiveness.

In addition to the normal Master Section plug-ins, various specialist plug-in and processes are on offer in the batch section, including trimmer and resizer tools, a DC remover, and the rather clever Meta-Normalizer, which lets you normalise a batch of files to the same loudness. Overall, batch processing is significantly easier to set up and use in this latest incarnation, and, let's face it, the easier a tool is to use, the more likely you are to use it.

Final Thoughts

I wasn't expecting this seventh Wavelab iteration to provide such radical benefits, but having used it extensively for various projects over several weeks, I'm loving all the improvements, and fully expect other existing users to do so. With such a major revision, there are inevitably a few teething troubles, including graphic bugs with various plug-ins (particularly on Macs). However, Philippe is already working on a 7.01 update to deal with those that are due to Wavelab 7 (others will have to be resolved by their respective developers), with a projected release date around mid-November, so that may already be out by the time you read this. [*Tests at the SOS office suggested that there are no major discrepancies between the Mac and PC versions — Ed.*]

A few Wavelab 6 PC features have been casualties in the cross-platform rationalisation, including the backup plan, ultra-safe (but slow) CD extraction, CD label designer, Audio Database, Montage video track and CPU overhead monitoring; this may annoy some upgraders, but some of these features may apparently reappear in future updates if users insist. Some PC users will also be disappointed that a printed manual is no longer included, as it was with all previous Wavelab versions, while new Mac users may end up lost on what is after all a totally new application to them. However, I'm one of those who really appreciates Wavelab 7's extensive and context-specific online help, which provides instant feedback on any function or feature when you really need it — ie. when you're about to use it. Steinberg also offer four online 'what's new' video tutorials (www.steinberg.net/en/products/wavelab/whats_new.html) that show most of the old and new features in action and provide a good overview for newcomers.

The full version of Wavelab 7 is expensive, but offers excellent value for money, while those on smaller budgets can investigate the cheaper but somewhat cut-down feature set of Wavelab Elements 7. However, for those who already own Wavelab 6, the upgrade price is an absolute bargain. I expected Wavelab 7 to be a good upgrade, but not this good! ■■■

£ Steinberg Wavelab 7, £497; upgrade from Wavelab 6, £82 or 29 Euros if purchased since April 2010. Prices include VAT.
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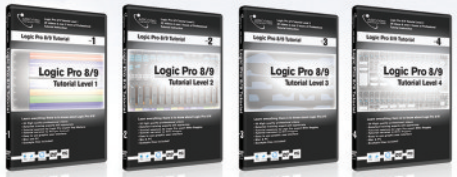

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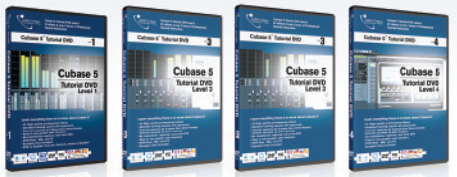

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The Pet Shop Boys: Chris Lowe (left) and Neil Tennant.
Photo: Ian Hooton/Retna UK

Classic TRACKS

Pet Shop Boys 'It's A Sin'

RICHARD BUSKIN

Protests against Catholicism have taken many forms, Martin Luther nailing his objections to the cathedral door, but the Pet Shop Boys chose to make theirs in disco...

It was the mid-'80s, synth pop was at its height, and in the process of creating a song with Chris Lowe that would subsequently mesh orchestral stabs, layers of keyboards, tons of echo, and assorted samples of Latin masses into one of the genre's most overblown, theatrically dramatic, disco-oriented masterpieces, Neil Tennant vented against the conflict between guilt and desire engendered by his Catholic upbringing.

"At school they taught me how to be," he wrote poetically of his education at St Cuthbert's High School in Newcastle upon Tyne, "So pure in thought and word and deed, They didn't quite succeed. For everything I long to do, No matter when or where or who, Has one thing in common, too. It's a, it's a, it's a, it's a sin..."

Featuring a characteristically thin, coolly dispassionate Tennant lead vocal set against the backdrop of Lowe's splashy melodic mélange, 'It's A Sin' was the

second Pet Shop Boys chart-topper in the UK and the best-selling European single of 1987, hitting number one in more than half a dozen countries and also making the top 10 in the United States. This, however, was a full three years after Britain's most commercially successful musical duo had first demo'd the track with Bobby Orlando (aka Bobby O). This Hi-NRG New York producer helmed their earliest recordings, before they severed contractual ties with him, to sign with EMI's Parlophone label in March 1985.

"That track had a great tune," says Julian Mendelsohn, who produced and engineered 'It's A Sin', as well as half of the *Actually* album that helped earn him a 'Producer Of The Year' nomination at the 1988 British Phonographic Industry awards. "If you don't have a great tune, you're stuffed."

Sarm East

Having acquired a love of recording technology in his native Melbourne, Australia during the late '60s, Mendelsohn relocated to England just after leaving school in 1971. This wasn't out of choice — the move was necessitated by his widowed mother's marriage to a British diplomat — but within a couple of years, after recording voice-overs and background music for a company that produced audio-visuals for conferences, he landed an engineering job at the small Milner Sound studio on London's Fulham Road. It was there that owner John Milner, an ex-BBC engineer, taught Mendelsohn the basics of mic placement, multitrack recording and tape editing. Following a five-year stint that saw him work with anyone from actor Michael Hordern to vocalists Linda Lewis and Paul Rodgers, the Aussie ex-pat made the transition from eight-track to 48-track in one fell swoop, when Jill Sinclair asked him to join the engineering team at her Sarm East facility in London's East End.

Initially working alongside Gary Langan as a tape-op/tea-boy, Mendelsohn subsequently earned his recording andmixing stripes on projects with Yes, the Buggles, Nik Kershaw, Tracey Ullman, Bronski Beat, Bob Marley, Musical Youth, Frankie Goes To Hollywood, Go West, Kate Bush, ABC and Level 42.

"At Sarm, Gary and I were on the »

Artist: Pet Shop Boys
Track: 'It's A Sin'
Label: Parlophone
Released: 1987
Producer: Julian Mendelsohn
Engineers: Julian Mendelsohn, Stephen Hague
Studio: Sarm West

» cutting edge of all the new equipment," Mendelsohn recalls. "Initially, the console was a Trident TSM and it was the first 48-track studio in Europe with two Studer A80s. There was also the Fairlight, the Linn Drum, the Synclavier and all of this other state-of-the-art gear, and when the 48-channel SSL arrived in 1982 it was only the second or third of those desks in the UK. It was a brilliant place to work."

Sarm West Studios, Notting Hill, London.

And it was also a great place to observe and learn from the contrasting production methods of Trevor Horn and Peter Collins.

"Trevor would usually end up taking a long, complex route to get the desired result, whereas Peter would usually take the quickest and most efficient way," Mendelsohn explains. "Peter would work from 10 in the morning until eight at night, Trevor would work from 11 in the morning until 11 in the morning, and both of them got fantastic results and were huge influences on me."

Although more in tune with Peter Collins' style of working, Mendelsohn would find himself caught up in the Trevor Horn approach on 'It's A Sin', but not before he'd already gone freelance and been cajoled into producing and engineering the hit track 'Suburbia' from the first Pet Shop Boys album, *Please*, in 1985.

"I didn't think their music would be my cup of tea, so at first I was a little reluctant," he now recalls. "However, Jill Sinclair, who was managing me, pushed me into doing it, and it was an absolutely fantastic experience. They were great to work with. They'd come up with terrific material and some really good ideas, and the sound was totally theirs — that was them telling other people how they wanted to sound and they were also responsible for all the arrangements. Chris was the one who came up with the hooks — the little synthesizer parts that you always remember — and Neil was really good in terms of the arrangement. They were very talented."

Sarm West

While Stephen Hague produced and engineered most of *Please*, he did the same for only a few of the 10 tracks on *Actually*. One of them was 'What Have I Done To Deserve This?' which featured Neil Tennant duetting with soul icon Dusty Springfield on a number about the mundane lives of bored '80s yuppies; 'Heart' was helmed by Andy Richards and mixed by Julian Mendelsohn; and Mendelsohn produced and engineered 'One More Chance', 'Shopping', 'Rent', 'Hit Music' and 'It's A Sin'. While 'What Have I Done To Deserve This?' was mixed by Mendelsohn, Hague ended up remixing 'It's A Sin'.

"We swapped our percentages," Mendelsohn says with a laugh. "There was an agreement."

The venue for all this work was Sarm West's Studio 2, which housed a 48-channel E-Series SSL and a Mitsubishi X850 digital 32-track. And in line with his preferred method of working, Mendelsohn had the Pet Shop Boys and their programming colleagues — including Richards, JJ Jeczalik and Gary Maughan — record the songs piecemeal. That way, they could constantly switch between tracks rather than complete one before commencing another.

"We never laboured on any track for more than a few hours," he says. "Otherwise, we'd just lose perspective. We wanted to keep everything fresh, and that was the way I generally liked to work. It was the Peter Collins approach: record, record, record. Most of it would be good, some might have to be done again. The only exception was when we were on a mission for a particular song part and might spend the whole day on it."

A case in point was 'It's A Sin', which, according to Mendelsohn, involved so much input on his part that he lost his objectivity on a couple of occasions and also lost his way.

"I remember describing a sound or a feel that I was after and Andy not quite understanding what I wanted," he says, referring to Andy Richards, who was the Fairlight and keyboard programmer on several of the tracks. "We were still there at midnight and he was saying, 'Well, what do you want?' My answer was, 'I want it like this, but I can't quite explain,' and there were a couple of times when we got fairly angry and frustrated with each other. I had this thing in my mind: 'I want it like that and I'm going to get it like that.'"

The control room in Sarm West's Studio 2.

Credit Where Credit's Due

The 1984 sub-six-minute demo of the song, as produced by Bobby Orlando, is similar to the finished record in terms of »


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» melody, tempo, and even Neil Tennant's vocal, but it lacks the atmospherics and drama of the released version: no NASA countdown, no choral chanting, no Latin mass, no cracks of thunder. Still, it was more than enough to serve as a template, and when work commenced on this at Sarm West, the track's general framework was quickly established.

Neil Tennant and Chris Lowe in Sarm West's Studio 2 control room during the recording of *Actually*.

"We started with the bass line and drum pattern and then began adding keyboards," Julian Mendelsohn recalls. "A lot of that was Fairlight, some of it was Roland rackmounted units, and after we recorded a guide vocal with a Neumann U87 we just built the track up from there. At one point, I remember taking a Nagra to Brompton Oratory [*nickname of the Church Of The Immaculate Heart Of Mary on Brompton Road in South Kensington*]. So that may be where we recorded the Catholic mass.

"The whole overblown production style was them telling me they wanted it overblown. And I was quite good at being overblown at that time. So was Andy Richards, who could be quite over the top. He did a lot of the programming on [*Frankie Goes To Hollywood's*] 'Relax' and a bit on 'Two Tribes', so there was a fair bit of Andy in 'It's A Sin'. He created that thunder sound on a Roland keyboard, and as things kept getting added, the two of us got quite carried away. There was no game plan; we just tried things, and if they worked, they worked, and if they didn't, we chucked them.

"My role as a producer on that album was just getting it all recorded. I also made a few suggestions here and there — you know, 'We can get a better sound for this' or 'Try singing that part again.' In those days, Neil wasn't a great singer and he knew it. So, there was always a bit of work involved in getting his vocals right. He's improved a hell of a lot since then.

"I always recorded things the way I wanted them to sound, so I never changed much in the mix. I mean, if you're not recording something the way you want it to sound in the end, what are you doing? You might as well get it right to start with. Neil and Chris would normally leave the studio at around seven or eight o'clock every night, giving us a list of things they'd like us to do, and then they would come back the next morning and tell us what they did and didn't like. So, that's how we'd develop a track, adding things all the time, and we would know when we had enough. In fact, we'd often have too much, which is why we would also dump some stuff.

"The five tracks I did on that album took a total of about eight or nine weeks to record and mix, and after that I had to leave for a project with Level 42. Then, when I returned two or three weeks later, I was told that Stephen [*Hague*] had been given a shot at mixing 'It's A Sin' and that was the mix they were going to use. He did a couple of neat little things, including dropping half a bar halfway through each verse to create a turnaround effect. I know that wasn't in the mix I did, but none of that bothered me. What was good was good. Some engineers used to get really upset when other people mixed their stuff, but I wasn't like that. All that mattered was everybody being happy with the end result.

"Still, even though Stephen's mix was the one that became a hit and was a bit more clever than mine, it was my one that actually sounded better. You could hear separate things jumping out at you and it had greater dynamics and a fatter sound, but it also got a bit boring from about halfway through the track and didn't maintain the interest as much as Stephen's mix. By bringing a pair of fresh ears to the proceedings, he was able to zero in on what the track needed.

"The exact opposite happened with 'What Have I Done to Deserve This?' Stephen had been working on the track for a week or two and was bored with it, and my fresh ears did the trick. You see, even though Dusty was a great singer, she was very long-winded when it came to getting the vocals right to her own satisfaction. In fact, when we did 'Nothing Has Been Proved' with her for *Scandal*, the film about the Profumo Affair, it really was like pulling teeth. She actually had a plan as to how she was going to sing the song from beginning to end — it wasn't random — but she'd only make it so far and, if she didn't get the last couple of words right, she'd say, 'No, I've got to start again,' because she wanted to do it all in one go.

King Controversy

Following the June 15th, 1987 release of 'It's A Sin' and the single's subsequent three-week stay atop the UK chart, DJ, producer and TV presenter Jonathan King used his regular column in *The Sun* to accuse the Pet Shop Boys of having crafted their latest hit by stealing the melody from Cat Stevens' 1971 recording of 'Wild World'. Then, never one to do anything by halves, especially when he could cause controversy and attract attention to himself in the full glare of the public spotlight, King tried to substantiate his claim by recording and releasing his own cover version of 'Wild World'. This utilised a similar arrangement to 'It's A Sin' but, despite his best efforts to discredit the electronic dance-music duo, King's single was a flop.

Evidently, no one really cared... that was, aside from the men actually being accused of the plagiarism. After suing Jonathan King, the Pet Shop Boys won out-of-court damages and donated the proceeds to charity.

Julian Mendelsohn (left) and assistant at Sarm West, 1987.

"That's how it was every step of the way, and I remember Neil and I looking at each other as if to say, 'Christ, this is going to take forever.' And it did take forever. We ended up having to sift our way through 20 tracks of vocals, but we got a fantastic result in the end, at which point we looked at each other as if to say, 'Well, that's why she took so long.' I had already worked with Paul Rodgers, who was one of the greatest rock singers in the world, and Dusty Springfield was another one. She was incredible."

Actually...

Following the November 1986 release of *Disco*, a collection of remixes of tracks from *Please* and the respective B-sides, *Actually* was the third Pet Shop Boys album but only the second to feature entirely new material. Issued in September of 1987, it sold over four million copies and spawned four UK top 10 singles: 'It's A Sin', 'Rent', 'What Have I Done To Deserve This?' — which, peaking at number two in both the UK and the US, was the biggest hit of Dusty Springfield's career — and a remixed version of 'Heart' that turned into another UK chart-topper.

"For the first year after I finish a record, I'll often listen to it and think, 'Oh God, why did I do this bit here? I should have done that,' but then I'll forget about all those little things that really wouldn't have made any difference," says Julian Mendelsohn. After producing more Pet Shop Boys material — including the hit cover of 'Always On My Mind' — he mixed 1990's *Behaviour* album and worked with the likes of Paul McCartney, Tasmin Archer, INXS, Fine Young Cannibals, Aztec Camera, Del Amitri, ABC, Go West, Kate Bush, Simple Minds and Liza Minnelli (who he co-produced with the Pet Shop Boys). Having relocated back to his hometown of Melbourne in 2005, he is currently recording a second album with New Zealand band the Glorious and doing work for various other local artists.

"Neil Tennant and Chris Lowe were great people to work with," he continues. "Pretty easy going, no stress, lots of ideas, always great tunes. What more can you ask for? If you're an engineer, it's the material you're given that supplies you with a good or bad sound. Of course, if you're a producer like Trevor Horn, you can take anything and mould it into something fantastic, but for the engineer it's what you're given from the other side of the microphone that serves as the basis of what you're going to end up with, and in that regard the Pet Shop Boys' material and performances of that material was hard to beat."

Melodyne: The Next Decade

Peter Neubäcker: Inside The Celemony Laboratory

MIKE SENIOR

It's 10 years since the groundbreaking Melodyne audio-processing software was launched. To celebrate, its inventor Peter Neubäcker invited us into his research lab to talk about possible future developments.

It has been a decade since the company Celemony was founded, and there's certainly every reason for them to celebrate this year on the occasion of their 10th anniversary. Not only have they consistently led the field in the field of monophonic pitch and time manipulation with the first three versions of their Melodyne software, but they've also rewritten the studio rulebook with their Direct Note Access (DNA) algorithm. This can manipulate individual notes within a mixed polyphonic audio file — a feat that even now remains out of reach of any of their commercial competitors.

The main technological brains behind this success are those of Peter Neubäcker, who I recently took the opportunity to interview in his home town of Munich as part of the company festivities. Surveying this softly spoken man's CV, it's fair to say that the average careers advisor wouldn't exactly have singled him out as a successful IT entrepreneur: a sometime hippie, luthier, and astrologer, by the early '80s Neubäcker had already been thrown out of school for playing truant, narrowly escaped six months in prison for refusing to do his compulsory military/community service, and built an alchemical laboratory in his basement!

Yet geometry and music appear to have threaded their way continually through his life, finally finding a focus in the study of harmonics, a subject examining numerical and musical relationships and championed by the Viennese Professor Rudolf Haase. It was Neubäcker's research interest in harmonics that first led him to experiment with computers, first the Atari, and then with CSound on the NeXT computer platform.

During the mid-'90s, he met Celemony's fellow-founder (and now Technical Director) Carsten Gehle through their common interest in NeXT programming, and when the idea of Melodyne hit, the two of them decided to put their heads together to bring

a product to market. "I had no experience whatsoever in the field of professional software development, and had totally underestimated the amount of work involved in going from a functioning prototype to an actual product," admits Neubäcker candidly, "[*so*] I would have failed hopelessly in this endeavour had it not been for [*Carsten*] and his gift for structured thinking and solidly based software design."

From then on, most studio jockeys are familiar with Celemony's rise and rise. First there was the original stand-alone Melodyne in 2001; then two subsequent versions that integrated with standard MIDI + Audio sequencers via the Melodyne Bridge plug-in; then a bona fide, all-in-one Melodyne plug-in; and finally the release, in late 2009, of the DNA-powered Melodyne Editor plug-in.

In Search Of DNA

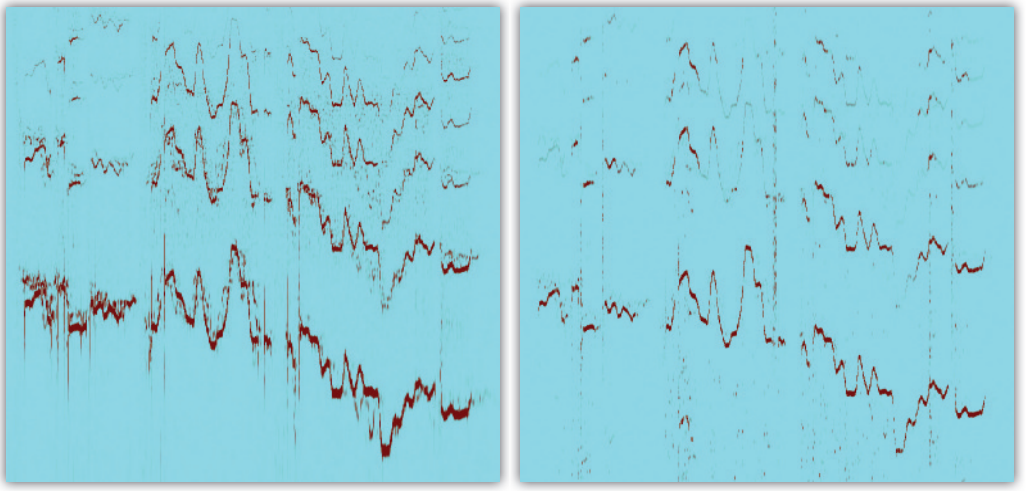
Neubäcker has always been very forthcoming in explaining how his processing algorithms work, and has already said much about the way the original Melodyne algorithm works: roughly speaking, it isolates periodic waveform elements to create a series of stand-alone snapshots of the audio signal (a bit like synthesizer wavetables), each of which represents the 'local sound' at a particular point in time. What quickly becomes clear as our conversation gets under way, however, is that the new polyphonic Melodyne algorithm isn't just an extrapolation on that approach, but had to be developed from scratch. "It's impossible to have the original idea of the 'local sound' in polyphonic mode," explains Neubäcker. "For the original Melodyne, I made the detection only in the time domain, not in the frequency domain, because I looked for periodicities. For polyphonic material, that's not possible any more, because there's no periodicity. It's not that simple. Yes, you can shift polyphonic material *en masse* with existing techniques, such as phase vocoding, but when it comes to separating out single notes from a polyphonic source, you have a mix of overtones and no means of finding out which overtone belongs to which note — in other words, which overtone is a fundamental and which is a harmonic. The greatest difficulty is to assign one part of an overtone to one note, and another part to another."

Even in the face of scepticism from many of his fellow software developers, Neubäcker nonetheless decided to actively hunt for a true polyphonic algorithm, and although progress turned out to be quicker than he'd expected, it was nonetheless still quite a long process. "But not of coding, just of thinking," stresses Neubäcker. "I was thinking, if a Fourier Transform gives me the signal's spectral components, and if we, as listeners, can hear what is in there, then there must be a way to separate them, if not totally, then at least to that degree that we can get control over the important parts."

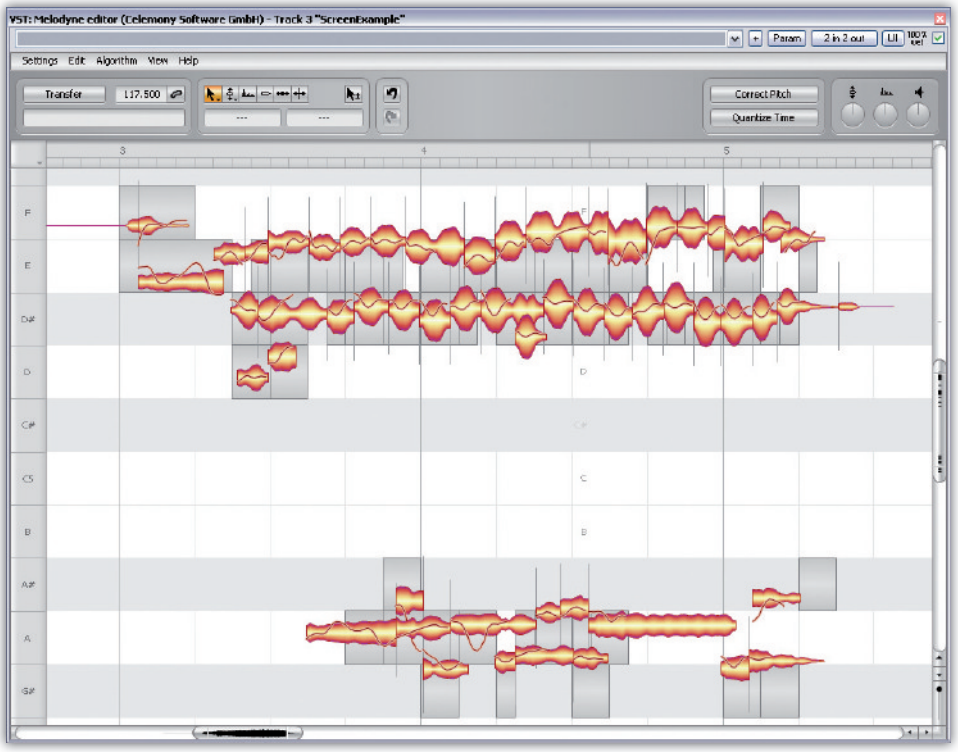
An important factor in solving this problem was the development of a method of detecting the notes within the audio, because the relevance of the different signal overtones could then be rated in relation to those notes. "On a spectrogram I see what I see, and I can interpret it in a way, but to make the program find what I see, I have to find a way to tell it that 'this is an object', 'this is a note', so that then the most relevant parts can be followed. The most difficult part of the work is to find out what is relevant and what is noise, because there may be musical notes that are very relevant, but quieter than any noise. There are things that we hear that are on a higher level of importance, and there are things that are not. If there's a little harmonic somewhere that's hardly audible »

Melodyne & Resynthesis

Peter Neubäcker: "In the original Melodyne, the formants were not analysed in terms of individual harmonics, but more in terms of the overall spectral shape. For example, if you retuned the formant in the original Melodyne, you shifted the whole formant spectrum. What we can do now with the new analysis is reshape the spectrum as well. I've done a lot of experimental stuff with it. For example, usually if you have a single monophonic guitar note, Melodyne Editor will select Melodic mode and group all the harmonics together. However, you can get into the field of creative abuse if you manually switch the detection mode to Polyphonic and then label some overtones as notes in themselves — if you retune them, you can get something more like a gong sound than a guitar. You can take a real flute recording, and make it sound more like a synthesizer, but because the original phrasing of the flute is always there, it's more alive than a synthesizer would be. You can turn any sound into a synthesized sound."



Here are a couple of spectrograms from the analysis software in Peter Neubäcker's own laboratory. The left one shows a section of one of the microphone signals from a unison boys' choir recording in which each of six choristers had his own mic. Notice the complicated overlapping spill signals, which can make note detection very unreliable. By comparing the different microphone signals, however, Neubäcker's prototype 'relevance evaluation' technology can much more reliably separate the wanted signal from the spill for note-detection purposes, as you can see in the right spectrogram.



Peter Neubäcker is working on improving Melodyne Editor's DNA note detection, which has trouble dealing with wide vibrato in polyphonic audio. In this example, strings of spurious notes are detected in polyphonic audio consisting of two very simple operatic female vocal phrases. While it's already fairly simple to clean up unwanted notes manually, the aim is for Melodyne to get much closer to the desired result automatically.

» and you can't decide if it belongs to this note or to that note, it really doesn't matter, because it's not so important."

The Problem Of Spill

The issue of 'relevance' continues to exercise Peter Neubäcker as he continues his research work. "What I'm interested in at the moment is processing multi-miked ensemble recordings. When recording a band live, you have all the individual instruments with a lot of spill between them. If you wanted to retune one instrument, you'd have to retune the spill in every other signal. I worked on the last Peter Gabriel CD, for example, which had a horn section on it that they wanted to retune. It was a very typical situation: they had five horns and each one had a microphone of its own, but there was a lot of spill, of course. If you retuned any individual mic, you'd have had to manually retune that spill on the other instruments by the same amount.

"My idea is that if you analyse each different microphone, the system can work out which is the main voice in that one, and handle the other signals appropriately. It doesn't remove the spill — it only does it virtually for the user. They'll only see the saxophone voice they're processing, and won't be seeing the spill, because that will be handled automatically. You just grab that instrument and move it around, and the system knows which spill notes on the other microphones have to be adjusted. But it's at the laboratory stage at the moment — what we had to do with this Peter Gabriel recording was mix it down to a stereo file to retune it, which meant that they didn't have to retune every file separately. But that meant they couldn't then re-use the individual files for a 5.1 mix or whatever."

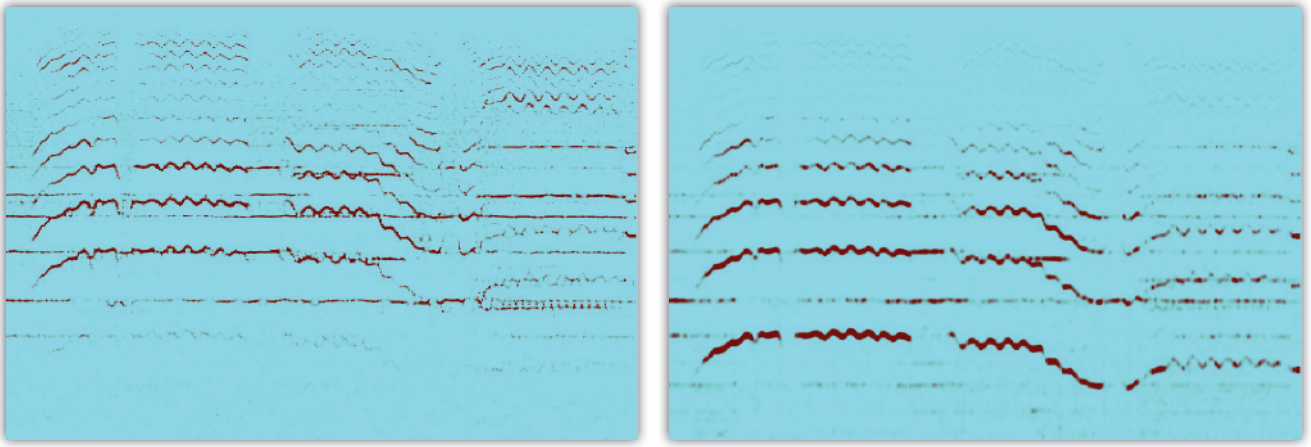
In discussing his ideas for dealing with spill, Neubäcker played me a few examples he'd been using in his research, and already at its current stage of development it appeared to be distinguishing between spill and non-spill signals very effectively. The most impressive example was where six young choristers singing in unison had each had their own mic, and the algorithm seemed able to distinguish between their very similar on-mic and off-mic sounds remarkably accurately.

"Of course this couldn't be in Melodyne Editor, because it's just one track," adds Neubäcker, "so it would have to be in a later version. An important aim for Melodyne Studio in the future is that there are tracks that 'know' each other, and know their context. In that future version there would be 'group tracks', where you would have to specify that certain recordings belong together, so that they are analysed together."

Developments In Note Detection & Noise Handling

Additional research is also going into improving the software's polyphonic note-detection process in response to more expressive internal lines. "If you have a better detection of the pitch curves," says Neubäcker, "the processing of the sound will be better as well. A pitch course with a lot of vibrato or portamento is followed perfectly well by Melodyne's monophonic algorithm, but the DNA technology doesn't yet handle this kind of line very well within a polyphonic context — the vibrato comes out as lots of different notes. I'm on the way to following the paths of the single instruments now, though, so if there are parts to be found, then that part of the detection will become a lot better. So in future there will be less to do in the Correct Detection mode, because I'm always working on how things can be assigned more plausibly in terms of which are notes and which are not notes."

It appears that detection may also improve in situations where band-limited signals have little, if any, fundamental frequencies for their lowest notes. One example Neubäcker played for me was an ancient recording of Caruso warbling over a full orchestra. Even though the recording included hardly any note fundamental at all for Caruso's voice, Neubäcker's improved 'relevance evaluation' routine was nonetheless able to correctly interpret where the note fundamental should have been. "You hardly have any fundamental at all — you only hear the overtones," he commented. "If I switch on the relevance evaluation, however, you suddenly see it down there. This isn't really in the signal, it's just reconstructed from the overtones. When the path-following takes place, it will find



Here are another couple of spectrograms from Peter Neubäcker's laboratory, which demonstrate the potential of his new 'relevance evaluation' algorithm. The left-hand spectrogram shows a raw analysis of an old Caruso recording, in which you can clearly see the pitch contour of Caruso's voice, but with a vanishingly low level of each note's fundamental frequency. The spectrogram on the right shows how the relevance evaluation can nonetheless reliably identify where the fundamentals lie.

these things down there as well."

When I reviewed Melodyne Editor back in *SOS* December 2009, one of my main criticisms was that Melodyne still struggled to cope with the less predictable noisy elements of musical signals — instrument transients, for example, often seemed to respond unpredictably when you manipulated note timings. Neubäcker suggests that additional control may be forthcoming in this department, via independent manipulation of what he calls the 'remainder signal', in other words the transients and other noises that remain when all the pitched information is removed: "After we introduced DNA processing, more than two years ago, we were not quite sure what the interace would be, and in fact we reduced some of the functionality because we saw that it was a mess to deal with everything, and we needed to gain some more experience of how people would wanted to work with it. Therefore we assigned the remainder signal to the notes that were detected, so that the user didn't have to deal with it. However, if we later introduce another processing level for this, things like a door slam or a click in the audio would get into the remainder track, so you'd have more objects to touch, so that you could say 'I don't want this object' or 'I want it louder', or something."

Fun as it is to gaze into the Celemony crystal ball, however, Peter Neubäcker is keen to point out that there can be no guarantee of when (or indeed if!) the fruits of his latest research will make their way into new versions of Melodyne. "I'm learning as well," laughs Neubäcker. "We had a very hard time after we announced DNA, and it took us nearly two years to get to the point of releasing Melodyne Editor. What is going to work and what is not going to work is still in the process. I'm in the happy situation at the moment where I'm more in the research laboratory and other people in the company are working on the concrete products, but we have so many ideas in our research laboratory in different stages of maturity that it's impossible to say with any confidence whether a certain feature will be available in a given version of Melodyne. I like to be very open about what I do, but I just want to prevent people getting expectations about what is to come in case it doesn't come, or doesn't come so soon!"

Perform First, Compose Later

Despite working for more than a decade on Melodyne's core technology, Peter Neubäcker continues to show an unswerving enthusiasm for what he's doing, driven by his desire to remove technological obstacles from the creative process. "The potential of this technology is incredible. I've always thought that if you can get access to every musical element, then it melts together the processes of music production and composition so that it's not important that you have the composition ready in the first place and then record it. You can record something first, and then see what you can compose with it.

"The problem with recording is that music is frozen at a certain time, and there's a very decisive separation of roles between the composer and the performer. We can change this if we give the instrumentalist, or the vocalist, or the musician the freedom to improvise or to be a composer after the performance — to re-perform the music according to their vision. Maybe the composer has a vision of his music (and I think it's the vision of this music that really matters) that no performer is able to perform? Or maybe some other composer afterwards says 'Oh, it would have been nicer this way'? The musician will have to be less of a technician, and more of a musician. As long as the technology serves the quality of the music, I think it can only make things better."

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Melodyne: November 2001
W www.soundonsound.com/sos/nov01/articles/melodyne.asp

Melodyne 2, featuring Melodyne Bridge: January 2004
W www.soundonsound.com/sos/jan04/articles/melodyne2.htm

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Melodyne Editor Plug-in: December 2009
W www.soundonsound.com/sos/dec09/articles/melodynedna.htm

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Records WAV, BWF and MP3 to Compact Flash cards or Micro Drives, optimized gain adjustment, 48V phantom power, analog limiter, USB 2.0 connection, balanced line ins, built-in high-fidelity microphone preamps, S/PDIF input, lithium-ion battery (recharge via USB).
€ 155.- £ 135.-
order code 184184

Zoom H2 4GB Bundle
Mobile recorder
24bit/96kHz, USB interface, record 360° sound as 2 channel data or 4 channel data simultaneously, mic/line input, headphone output, perfect for interviews, podcasts, meetings and live recording, incl. USB cable, stand, power supply, windscreen and 1GB SD card. **Bundle includes additional 4GB SD card.**
€ 175.- £ 153.-
order code 203048

Olympus LS-5
Portable digital recorder
24bit/96kHz, WAV, WMA, MP3 and PCM recording, built-in microphone, large back-lit display, 2GB internal memory, SD/SDHC card extension slot, internal stereo speaker, USB 2.0 connection, optional remote control, dimensions WHD: 4,8 x 13,15 x 2,24cm, weight: 165g (incl. battery).
€ 179.- £ 156.-
order code 250006

Edirol R-09 HR 4GB SD Card Bundle
24bit/96kHz mp3/WAV recorder
Up to 320kbps mp3 playback and recording, recording on SD or SDHC card (up to 8GB), integrated stereo condenser mic, OLED display, built-in preview speaker, incl. wireless remote control with split function, USB 2.0, stereo mic in, stereo line in, Incl. power supply, 512MB SD card, USB cable, Cakewalk Pyro AudioCreator LE, wireless remote controller and small table mounting plate. Bundle incl. 4GB SD card.
€ 279.- £ 244.-
o. code 207377

Sony PCM-M10
Portable 24bit/96kHz audio recorder
Built-in stereo microphone, internal speaker, cross-memory recording, internal 4GB flash memory + MicroSD/memory stick micro slot, limiter and low-cut filtre, 5s. pre-record buffer, manual or automatic recording level control, USB-2.0 connection, includes power supply, Soundforge Audio Studio LE, cable remote, 2 AA batteries and cable, finish: grey.
€ 325.- £ 284.-
o. code 238886

Zoom H4 N Wizoo Bundle
24bit/96kHz digital recorder
Perfect for interviews, podcasts, meetings and live recordings. 2 built-in stereo microphones, requires 24V or 48V phantom power, USB port. Includes windscreen, USB cable, cover and Cubase LE4. **Bundle includes 4GB SD card, Superlux HD681 stereo headphone and Wizoo Publishing ‚Mobile Recording' Thomann Special Edition by Kai Schwirzke.**
€ 339.- £ 296.-
o. code 232016

Zoom R16 Bundle
USB audio interface and DAW controller
16-track playback and 8-tracks recording, 24bit/48KHz PCM recording, internal stereo condenser mic, 8 mic ins, 2 outs, 8 balanced XLR / 1/4" TRS connections, internal effects, Mackie Control emulation via USB, works as USB storage, support for Win XP/Vista and Mac OS, incl. Steinberg Cubase LE4 and 1GB SD card. **Bundle incl. the t.bone MLS66 stereo headphones and 4GB SD card.**
€ 355.- £ 310.-
order code 245233

Zoom R24 Bundle
Digital recorder
24-track playback and 8-track recording simultaneous, 24bit/96kHz, 8 ins and 2 outs, built-in stereo mic, compressor/EQ, tuner and metronome, support SDHC cards up to 32GB, USB 2.0, 8x XLR-/TR-Input Mic/Line/Hi-Z, headphone out, incl. Steinberg Cubase LE, 1GB SD card, USB Stick (2GB with Drum Loop Library), power supply and cable. **Bundle incl. the t.bone SC400 studio microphone, the t.bone HD-800 stereo headphones and cable.**
€ 515.- £ 450.-
o. code 253669

the t.mix Mix 502
5-channel mixer
1x mic in with 2-band EQ, 2x stereo ins, 2-track I/O, ¼" balanced jack out. Dimensions: 13.8 x 22 x 2.8-4.5cm. Weight: 0.8kg
€ 31.- £ 27.-
order code 207200

the t.mix Mix 802
2x mic ins with 3-band EQ, 2x stereo ins, 2-track I/O. Dimensions: 19.3 x 26.8 x 2.8-4.5cm. Weight: 1.2kg
€ 46.- £ 40.-
order code 207202

Alesis Multimix 4 USB
4-channel mixer with digital output
16bit, 44,1kHz signal on USB, 4 line level ins, 2 XLR ins with gains and switchable 48V phantom power, high impedance guitar input, 2 channel EQ on mic ins, multicolor LED metering, main and headphone outs with independent level controls, incl. power supply, dimensions WHD: 15,2 x 19,6 x 5cm, weight: 0,6kg.
€ 95.- £ 83.-
order code 235244

the t.mix 1832 FX
Rack mixer
18 channels, 6 microphone inputs, 4 stereo inputs, built-in USB/MP3 player, 24bit DSP effects unit with 256 programs, 9-band EQ, 3-band EQ mit patametric mids (channels 1-4), 4-band EQ (channels 5-12), 2-track I/O, 4 aux outputs, dimensions: 41,5 x 40 x 11,5cm, weight: 6,7kg.
€ 235.- £ 205.-
order code 242617

Allen & Heath ZED-10FX
Mixer
4 mic/line ins, 2 dual stereo ins, DuoPre™ preamp, 2 aux sends, USB send/return, internal 24bit FX processor, HP-filter, stereo return, 2-track return, main insert, internal power supply, weight: 3,3kg.
€ 275.- £ 240.-
o. code 246916

Allen & Heath ZED-10
without FX processor.
€ 189.- £ 165.-
order code 246915

Soundcraft EFX 8
8-channel mixing desk
8x mono inputs, 2x stereo inputs, inserts on all channels, 1x FX send, 1x aux send (pre/post), 48V phantom power, solo/mute switch, integrated Lexicon FX with 32 presets, peak LEDs on all channels, integrated power supply. Dimensions: 33 x 9.1 x 36.2cm. Weight: 4.6kg. Optional rackmount available.
€ 285.- £ 249.-
order code 118879

Behringer Xenyx X2442 USB
24-channel mixer with 60mm faders
10 mic ins with 48V phantom power, 75Hz low cut, compressor, 24bit multi FX with 16 presets, internal USB sound card, 4 stereo ins, 3-band EQ with parametric mids, 4 aux (pre/post), insert, Peak LED and mute each channel, 2-track I/O, XLR main out, 4 sub groups, internal power supply, includes 19" rack kit and energyXT2.5 compact music production software.
€ 299.- £ 261.-
o. code 242913

Yamaha MG 166cx
16 channel mixer
8x mono mic/line ins (XLR/jack), 2x mic/line ins (XLR/stereo jack), 2x stereo line ins (jack), 3x aux sends, 8x inserts, 3-band EQ with parametric mids, internal compressors per mic channel, SPX FX processor with 16 programs, 18dB high pass filter, high-end mic preamps, 48V phantom power. Dimensions: 47.8 x 10.2 x 48.9cm. Weight: 5.5kg
€ 358.- £ 313.-
order code 115511

Alesis MultiMix 16 FireWire
16ch. mixer with FireWire interface
8x mic/line ins, 4x balanced stereo line ins, 3-band EQ per channel, 2x aux send/returns, 100x 28bit FX programs, headphone out. FireWire interface: 18 ins, 2 outs, 24bit/48kHz, ASIO/WDM drivers for Win XP SP2, Core Audio drivers for Mac OS X, includes Cubase LE.
€ 389.- £ 340.-
order code 186079

Behringer SX2442FX
24-channel mixing desk
16x XLR mic ins, Xenyx mic preamps, 3-band-EQ, 2 xFX processors with 100 presets, stereo 9-band graphic EQ, FBQ feedback detection system, 4x subgroups, 4x auxes per channel, internal power supply.
€ 429.- £ 375.-
order code 217454

SX3242FX
32 channels.
€ 479.- £ 418.-
order code 217456

Mackie 1642 VLZ3
Compact 16-channel mixer
8x mono mic/line channels, 2x mic/line stereo channels, 2x stereo channels, 4x subgroups, 4x aux (2x pre/post switchable), XDR2 mic preamps, 60mm faders, 3-band parametric EQ (mono channels), 4-band EQ (stereo channels), 48V phantom power. Dimensions: 42.5 x 13.1 x 42.1cm (WxDxH). Weight: 8.3kg
€ 545.- £ 476.-
order code 112362

Yamaha MG 32/14FX
32-channel live mixer
24x mic ins, 4x stereo ins, 2x SPX-quality FX processors, 7-band master EQ, 2x aux sends, 2x stereo returns, 24x inserts, 3-band EQ with parametric mids,18dB low cut filter, 48V phantom power, internal power supply. Dimensions: 102.7 x 14 x 55.1cm. Weight: 20kg
€ 1066.- £ 931.-
order code 161994

Mackie Onyx 32.4
Live mixer
28x mono ins, 4x subgroups, 3x outs, 4-band EQ with fully parametric mids and fixed lo/hi, 30dB pad, low cut, 6x aux sends, LED and insert per channel, 6x2 matrix, internal compressor/limiter, D-Sub 25 direct out, talkback, phones out, main insert, internal power supply. Weight: 21.8kg
€ 1125.- £ 982.-
order code 194635

Phonic Summit
Digital mixer
16 channels, 24bit/96kHz, phantom power, 17x 100mm motor fader, touch screen, 26 semi-parametric 4-band-EQ for all inputs, AES/EBU I/O, 8 multi-function outs, 2 effect processors, 26 dynamic processors, SD card, optional Expansion Card compatible with Win XP/Vista/7 and Mac OSX (incl. Snow Leopard), ext. power supply, weight: 10,5kg, incl. 19" rackmounts.
€ 1777.- £ 1551.-
order code 246336

Presonus Studiolive 16.4.2
16-channel digital mixer
Built-in 32x18 FireWire audio interface, ‚Fat-Channel' processing with 4-band EQ, compressors, limiters and gates. 16x inputs, 6x auxiliary mixes, 4x subgroups, 16x Class A XMAX mic preamps, 44.1kHz & 48kHz sample rate, 2x master DSP FX, talkback. Dimensions: 43.7 x 17.5 x 56.8cm (WxHxD).
€ 2169.- £ 1894.-
order code 225534

Mackie DXB200 Refurbished Bundle
Digital open architecture mixing desk
72ch/96kHz (36ch/192kHz), 8 busses, 2 touch screens, 25 motorised faders, 4-band EQ, compressors and gates, VST plug-in compatible, 2 PCI slots, flexible I/O, 9 slots for optional I/O cards (not included in basic config), Mackie Control Universal mode for DAW control. **Bundle incl. Xbus AES card, Xbus digital card and 2x Xbus line card. B-Stock with full warranty!**
€ 3666.- £ 3200.-
order code 251867

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Fax: +49 9546 9223-24

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- free shipping to the UK and Ireland on orders above £159

Miditech Midistart Music 25
MIDI master keyboard
25 touch sensitive keys, joystick for pitchbend and modulation, octave up/down, MIDI output, USB connection, includes USB cable and Magix Samplitude SE, dimensions WHD: 38 x 8 x 24cm, weight: 2,3kg.

order code 230804 € 55.- £ 48.-

Miditech Midistart 3 USB
MIDI keyboard
49 full-size touch sensitive keys, pitch bend and modulation wheels, USB port for direct connection to Mac/PC, MIDI out, includes USB cable and Magix Samplitude SE PC software. Finish: Silver

order code 116317 € 65.- £ 57.-

Thomann SP-5500
Stage piano
88 hammer action keys, 559 sounds, 203 styles, 3 user styles, 100 songs, 64-note polyphony, accompainment control (start/stop, sync start/stop, fill-in A/B, fade), dual mode, split mode, DSP, transpose function, lesson function, pitch bend, headphone out, stereo aux I/O, MIDI I/O, USB port. Finish: Silver Grey

order code 154087 € 368.- £ 321.-

Thomann SPWS-5500
Wooden stand for SP-5500
order code 204071 € 46.- £ 40.-

Korg SP-170
Stage piano
88 natural weighted hammer action keys, 2x9W built-in speakers, 10 sounds, effects: reverb and chorus, key transpose and pitch control, 2 line/headphone outs, MIDI out, dimensions: 131 x 32.5 x 13.3cm, weight: 12,1kg.

Black
order code 242228 € 499.- £ 436.-
White
order code 242231 € 499.- £ 436.-

Yamaha P155
Stage Piano
88 graded hammer action keys, pure CFIII piano voice, 17 voices, polyphony 128 voices, dual and split mode, metronome, 2 headphone outs, 2x 12W, includes note stand, FC-4 pedal and PA301 power supply, weight: 18,6kg.

Black & ebony
order code 223258
Black & mahogany € 1135.- £ 991.-
order code 223244

Kawai ES 6
Stage piano
88 keys, 32 sounds,192-voice polyphony,100 rhythms, transpose, intonation, string response, brilliance, dual/split mode, 2-track recorder, metronome, 2 headphone outputs, MIDI I/O, line I/O, USB-host. Includes F10H sustain pedal, power supply and music rest.

Finish: Black
order code 218136
Finish: Silver € 1150.- £ 1004.-
order code 218137

Roland FP-4
Compact digital piano
Speaker system, 88 PHA Alpha II keys, 128 voices, 333 sounds and 9 drum sets, GM2 system, 3-track recorder, 1/4" jack I/O, 2x 1/4" TRS stereo headphone outs, USB, MIDI I/O, foot pedal inputs (damper, soft, sostenuto), DC in (9V AC power supply). Dimensions: 134.2 x 30.5 x 13.5cm (WxDxH). Weight: 15.3kg. Finish: Black

order code 112451 € 1169.- £ 1021.-

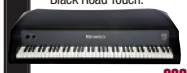
Clavia Nord Electro III
Stage keyboard
Based on authentic emulations of Hammond B3 and electric pianos, Nord sample library support, flash memory for 256MB samples and 126 programs, new effects and amp emulations, compressor and EQ.

61 waterfall keys € 1577.- £ 1377.-
order code 220551
73 waterfall keys € 1725.- £ 1506.-
order code 220582

Clavia Nord Stage 88 EX
Stage piano
88 weighted keys with aftertouch, 3 split zones, organ section with Hammond B3, Vox Continental and Farfisa organ, piano section with Grand Piano, Rhodes, Wurlitzer, Hohner Clavinet and many more, large effect FX, 256 MB memory for piano samples.

order code 219785 € 2959.- £ 2583.-

Rhodes Mark 7 73 S Black Road Touch
Analogue electric piano
73 keys, an extremely responsive mechanical action with the most professional touch, includes stand and pedal, dimensions WHD: 118.1 x 20.9 x 58.4cm, weight: 43.1kg, finish: Black Road Touch.



order code 244334 € 2999.- £ 2618.-

Doepfer Dark Energy DVD Bundle
Expander
Monophonic stand-alone analog synthesizer with USB and MIDI interface, USB, MIDI input, 4 CV outputs, 1 gate output, external power supply, dimensions WDH: 18,5 x 14,5 x 6,5cm, weight: 1,2kg. **Bundle including german video tutorial DVD „Hands on Synth Sound".**

order code 245106 € 375.- £ 327.-

Neo Instruments Ventilator
Rotary Cabinet Simulation
Reproduction of a Leslie 122 Roto effect, rotor speed adjustable, drive section for real tube feeling, variable mic position of the virtual mics, true bypass, speaker simulation disconnectible for guitar amp, connection for remote footswitch/halfmoon switch, stop function with remote switch, power supply included, dimensions: 19.2 x 15.3 x 5.5cm, weight: 1.1kg.

o. code 237621 € 377.- £ 329.-

Akai MPC 2500 SE
MIDI production workstation
Special edition with blue display, 128MB, 32-voice polyphony, 64 MIDI channels, 16 dynamic pads, 4 pad banks, Q-Link faders and knobs for real time control, optional hard drive, USB port, chop-shop loop-slicing, patched phrase samples, pad sample edit, 10 analog outs plus stereo digital I/O, 4 Q-link controller (2 faders and 2 knobs), S/PDIF coaxial I/O, includes 32MB compact flash card.

o. code 240640 € 1222.- £ 1067.-

Akai Miniak
Performance and production synthesizer
8 voices with 3 oscillators, up to 8 multi-timbral parts with stereo effects and 40-band vocoder, 2 multi-mode filters, 3 envelope generators, 2 LFOs, sample and hold, tracking, dynamic real-time and step sequencers, arpeggiator, and drum machine/rhythm, sequencer, 37-key synth-action keyboard with velocity sensitivity, includes VPM1 gooseneck microphone and power supply.

order code 228545 € 295.- £ 258.-

Yamaha Motif XS 7
88-key workstation
355MB Wave ROM, 128 voices, presets: 1024 normal voices + 64 drum kits, colour display, 4-part arpeggiator, 4 layers or splits in performance mode, sequencer, internal sampler, USB, Ethernet

order code 111124 € 2099.- £ 1832.-

Novation Launchpad
Ableton Live/MIDI controller
Bi-directional communication, 64-button grid with colour-synced status feedbac, scene control switchable to control volume, sends, muting and more, up to 6 units may be operated simultaneously, USB bus-powered, large rubber grips, automap-enabled, includes Ableton Live 8 Launchpad version. Weight: 0.76kg

order code 236641 € 175.- £ 153.-

Akai APC 20
Ableton Live USB controller
Exclusive bi-directional communication between the APC 20 and Ableton Live (no mapping required), USB plug-and-play, 40 triggers, 1 rotary control knob, 8 channel faders, 1 bus fader, compatible with PC and Mac, includes Ableton Live Lite Akai APC 20 edition.

order code 243147 € 185.- £ 162.-

Native Instruments Traktor Kontrol X1
USB DJ controller bag bundle
Instant recognition by Traktor, software controlled backlit buttons, fits perfectly alongside a standard club mixer, super-intuitive loop and cue control section, 8 knobs and 8 buttons, push encoders, expandable (connect two X1s to control four decks), incl. Traktor LE, Kore2 Player (incl. selection library) and MIDI templates for controlling other DJ and performance software. **Bundle incl. X1 Bag**

o. code 243915 € 199.- £ 174.-

Native Instruments Maschine
Groove production studio
Pad controller, 16 dynamic pads with aftertouch, 2 backlit displays, 11 rotary knobs, 41 buttons, USB-powered, MIDI I/O, pattern-based sequencer with 64 patterns per group, step sequence programming and realtime recording, drum grid and piano roll editors, sampler, > 20 built-in effects, >5GB sounds, PC/Mac.

order code 223060 € 499.- £ 436.-

Euphonix MC Control V2
DAW controller
Touch-sensitive knobs, motorised faders, touchscreen display and OLED, supports HUI and Mackie Control compatible DAW software for Mac OSX. Dimensions: 50 x 23.5 x 3cm. Weight: 2.5kg.

order code 248964 € 1399.- £ 1221.-

ESI DuaFire
24bit/96kHz FireWire audio interface
2x ins with level controls (¼" jack), mic preamps with phantom power (XLR), 2x hi-Z ins, 4x analogue outs, direct input monitoring with level control, headphone out, bus powered or via separate power supply. Supports DirectWIRE 3.0, MME, ASIO 2.0, GSIF 2.0, DirectSound and Core Audio. Includes power supply and Steinberg Cubase LE 4.

order code 206255 € 95.- £ 83.-

M-Audio Fast Track MKII
USB audio interface
24bit/48kHz, XLR microphone input with 48V phantom power, 1/4" instrument input with gain control, 1/4" stereo headphone output, RCA stereo speaker outputs, plug-and-play USB connection (USB 2.0 compatible), including Pro Tools M-Powered Essential.

order code 237376 € 118.- £ 103.-

M-Audio Fast Track Pro
USB audio interface
4x4 24bit/44.1kHz, stereo in up to 24bit/96kHz, USB powered, 2x mic/instrument preamps with switchable phantom power, 2x inserts, 2x balanced line outs, 4x additional RCA outs, S/PDIF coaxial I/O, S/PDIF out (AC-3/DTS surround sound compatible), MIDI I/O, compatible with both Mac and PC.

order code 184360 € 155.- £ 135.-

ESI ESP1010e
24bit/96kHz PCIexpress audio interface
PCIe card with external 19" interface, 8 analog ins (2 with mic preamp and 2 Hi-Z instrument), +48V phantom power, up to 8 analog ins and outs, coaxial S/PDIF I/O, 2 headphone outs, 2 MIDI I/Os, support for DirectWIRE 3.0, EWDM driver: MME, DirectSound, ASIO 2.0 support, Windows 7/Vista/XP/2000/2003 compatible, power supply optional, incl. Steinberg Cubase LE 4.

order code 230632 € 155.- £ 135.-

Digidesign MBox 2 Micro
Mobile Pro Tools LE system
About the size of a typical USB flash drive, high-quality sound, stereo out for headphone or speaker monitoring (no audio ins). Includes Pro Tools LE, Bomb Factory® and DigiRack™ plug-ins and Digidesign's Xpand!™ sample-playback/synthesis workstation, for Win XP and Mac OS X, USB bus powered. Work with Pro Tools and compose wherever inspiration takes you.

order code 139283 € 225.- £ 196.-

Focusrite Saffire Pro 24 DSP
FireWire audio interface
Virtual reference monitoring technology, real-time FX (EQ, compressor, comfort reverb), 24bit/96kHz, total I/O count of 16 ins (including internal loopback) and 8 outs, 2x mic preamps, 4x analog ins, ADAT optical I/O, 1x coax S/PDIF I/O, 2x headphone outs, MIDI I/O, 48V phantom power, LED meter, includes plug-in suite and power supply.

order code 233752 € 349.- £ 305.-

Presonus Firestudio Project
FireWire audio interface
24bit/96kHz, 8 XMAX Class A mic preamps, 8 analog mic/line ins, 2 instrument ins, 8 analog line outs, S/PDIF digital I/O, MIDI I/O, balanced send/return, zero latency monitoring with FireControl mixer/router, headphone output, 8 input level meters, includes recording software Presonus Studio One Artist, 19"/1U.

order code 199889 € 389.- £ 340.-

Apogee Duet Bundle
FireWire audio interface
24bit/96kHz, 2 channels, Firewire 400 I/O, breakout cable with 2 XLR mic ins, 2 1/4" jack instrument ins, 2 1/4" jack monitor outs, multi-segment LED display input and output levels, multi-function controller knob, headphone out, Maestro software for advanced control and low latency mixing, compatible with any Core Audio compliant audio application, compatible with Mac OS X Core Audio. **Bundle incl. original Apogee carry case and 2GB USB stick.**

o. code 209773 € 415.- £ 362.-

Focusrite Saffire PRO 40
24bit/96k FireWire audio interface
20x I/O, 8x mic preamps, 8x analog I/O (2x mic/line/instr. combo XLR, 6 mic/line combo XLR), ADAT I/O, 2x S/PDIF I/O, 2x monitor outs, monitor switch, 2 separate headphone buses, MIDI I/O, zero-latency DSP mixer/router, internal power supply, includes plug-in suite. Dimensions: 19"/1U. Weight: 3kg

order code 219725 € 458.- £ 400.-

MOTU Ultralite MKIII Hybrid
FireWire and USB audio interface
24bit/192kHz, 2 mic/instrument inps, 6 bal. ins and 10 outs (1/4" TRS), 48V phantom power, S/PDIF I/O, headphone out, CueMix FX, internal DSP, LCD display, suitable for use as a standalone mixer, compatible with Windows and Mac, supports WDM, ASIO and Core Audio, incl. AudioDesk software for Mac.

order code 239141 € 479.- £ 418.-

Focusrite Liquid Saffire 56
FireWire audio interface
24bit/192kHz with Liquid preamps, 28x I/O, 8x XLR mic ins, 2x ADAT I/O, coax S/PDIF I/O, 8x analog ins (¼" balanced jack), 10x analog outs (¼" balanced jack), Word Clock, MIDI I/O, includes Focusrite VST/AU plug-ins. Dimensions: 35 x 9 x 23.5cm (19"/2U). Weight: 5kg

order code 228525 € 679.- £ 593.-

RME Fireface 400
FireWire audio interface
24bit/192 kHz high performance FireWire audio interface, analog technology from ADI-8 converter, mic preamp technolgy from Quad and OctaMic (2 mic preamps), TotalMix technology from Hammerfall DSP series, very reliable drivers.

order code 193883 € 799.- £ 698.-

RME HDSPE Madiface
PCI express interface
MADI ExpressCard (34mm standard) for notebooks, 64 input/64 output channels, 1x MADI I/O (optical and coaxial), 44,1kHz/192kHz, clock-mode slave and master, sync sources: MADI coaxial, MADI optical and internal, connections: breakout box: input and output MADI optical, MADI coaxial (BNC), MIDI I/O, incl. DigiCheck software, ExpressCard MADI, 1m cable (IEEE 1394), driver-CD.

order code 213735 € 1299.- £ 1134.-

Digidesign Digi 003 Factory Complete
Complete Production Toolkit
18 simultaneous channels of audio I/O, 8 analogue ins and outs, 4 mic preamps, 48V phantom power, 8 motorized faders, 8 rotary encoders for pan/send/meter/plug-in control, 8-channel ADAT optical I/O or 2-channel S/PDIF optical I/O, 1x MIDI in, 2x MIDI out, BNC word clock I/O, Digidesign Pro Tools LE 8, Factory plug-in bundle and Complete Production Toolkit.

order code 232218 € 3055.- £ 2667.-

Native Instruments Komplete 7
Complete package of software
Contains: Kontakt 4.1, Absynth 5, Guitar Rig 4 Pro, Reaktor 5.5, Battery 3, FM8 and Massive, more than 10000 presets and sample-librarys with 100GB, all instruments as stand-alone or plug-in usable, supported formats: stand-alone, Audio Units, VST, RTAS, ASIO, Core Audio, Core MIDI, DirectSound.

order code 252237 € 459.- £ 401.-

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Sennheiser MKE 400
Microphone for video cameras
Hyper-cardioid polar pattern, frequency range: 40Hz - 20kHz, 3.5mm stereo jack plug connection, includes windshield, connection cable and 1x 1.5V battery (AAA). Size of base plate is 180 x 180mm.
order code 204069 — € 145.- £ 127.-

Rode NTG-2
Directional condenser microphone
Broadcast quality, condenser transducer, low-noise circuitry, low handling noise, phantom powered (44 - 52V) or battery powered (1.5V), 20Hz-20kHz, 131dB SPL max., ideal for film/TV/video. Includes stand mount, windshield and zip pouch. Weight: 161g
order code 179393 — € 185.- £ 162.-

Rode NT5-MP
Condenser microphone stereo bundle
2 matched small diaphragm condenser microphones, 0.5" gold sputtered capsule, cardioid, 20Hz-20kHz, excellent for drum overheads, acoustic guitar and stereo live recordings. Including plastic case, microphone clips and wind screens.
order code 154595 — € 269.- £ 235.-

Oktava MK012-01 MSP2 Matched Pair
Matched stereo set
2x small diaphragm condenser microphones, cardioid, 20Hz-20kHz, delivered in a wooden box. Finish: Silver. Made in Russia!
order code 165104 — € 279.- £ 244.-

MK012-01 MSP2
Single mic, cardioid
order code 165103 — € 149.- £ 130.-

Neumann KM 184 Stereo-Set
Small diaphragm condenser mic bundle
2x KM184 condenser microphones with sequential serial numbers, cardioid, 20Hz-20kHz. Includes 2x WNS100 windshield, 2x SG 21/17 stand adapters and wooden box.
Finish: Silver
o. code 153692 — € 1099.- £ 959.-
Finish: Black
o. code 158648

the t.bone SC440 USB
USB condenser studio microphone
USB connection for direct access to Mac and PC (not compatible with Windows Vista!), cardioid polar pattern, 20Hz-18kHz, includes plastic case.
order code 197603 — € 55.- £ 48.-

the t.bone SC450 Set
Large diaphragm studio microphone
Cardioid, external low cut and -10dB pad switch, 200Ω impedance, requires 48V phantom power, 30Hz-20kHz, includes shockmount and PVC case. Dimensions: 50.5 x 190mm. **Bundle includes the t.bone MS180 pop shield.**
order code 203194 — € 93.- £ 81.-

SC450 Stereo-Set - matched stereo pair in case (pop shield not incl.)
order code 174363 — € 169.- £ 148.-

StudioProjects B1
Large diaphragm condenser microphone
1" Capsule, transformerless circuit, requires +48V phantom power, cardioid, 20Hz-20kHz, -34dB sensitivity (0dB= 1V/Pa), 200Ω, 132dB maximum, 12dB-A noise (IEC651). Includes foam wind screen, zippered bag and shockmount.
order code 180703 — € 105.- £ 92.-

Rode NT1-A Complete Vocal Recording Solution
Large diaphragm microphone
20Hz-20kHz, 100Ω impedance, 132dB dynamic range, 137dB max SPL. Includes SM6 deluxe shockmount, 6m cable and Peter Freeman ‚Studio Secrets' tutorial DVD.
order code 235937 — € 179.- £ 156.-

AKG C3000
Studio large diaphragm condenser mic
Cardioid, 20Hz-20kHz, 200Ω, switchable -10dB pad, includes shockmount. Diameter: 53mm. Length: 162mm
order code 141408 — € 185.- £ 162.-

Rode T-1000 Thomann Edition
Large diaphragm condenser microphone
Identical as Rode NT-1000, transformerless SMT circuitry, 1" HF2 capsule with gold-plated diaphragm, incorporating internal shock mounting, 134dB dynamic range, 140dBA SPL capabilities, frequency response 20Hz - 20kHz, Cardioid polar pattern, Thomann Edition in durable satin black finish including shockmount SM6, cable and bag.
order code 247421 — € 211.- £ 184.-

the t.bone SCT2000 Set
Valve studio microphone
Truly warm sound, 20Hz-20kHz, omni/cardioid/fig-8 plus 6 intermediate stages, up to 130dB SPL. Includes case, shockmount and power supply. **Bundle includes the t.bone MS180 pop shield.**
order code 203196 — € 279.- £ 244.-

Rode NT2-A MS-180 Bundle
Large diaphragm microphone
Switchable polar pattern (omni, fig-8, cardioid), 1" diaphragm, 20Hz-20kHz, 147dB max SPL, includes SM2 shockmount. **Bundle includes the t.bone MS180 pop killer.**
order code 223002 — € 285.- £ 249.-

SE Electronics SE 2200T
Large diaphragm condenser mic
Gold plated 1" diaphragm, 100Hz switchable high-pass filter, -10dB switchable pad, frequency response: 20Hz - 20kHz, sensitivity: -33dB +1dB (0dB=1v/pa 1000KHz), impedance: <= 200Ω, equivalent noise level: 16dB (A weighted), max SPL: 130dB (0.5% THD@1000Hz), connector: 7-pin, power supply and shockmount included.
order code 244907 — € 339.- £ 296.-

Shure SM 7 B
Studio microphone
Dynamic studio microphone with cardioid polar pattern, 50Hz-20kHz, bass rolloff switch, mid-boost switch, 150?, shielded against broadband interference, fixed stand adapter, XLR connector, includes windscreen.
order code 129929 — € 385.- £ 336.-

EV RE20
Dynamic large diaphragm microphone
RE series, cardioid, switchable HP filter, variable D design, 150Ω, 45Hz-18kHz, includes clip and box, ideal for vocals, brass and bass drum. Length: 217mm. Diameter: 54mm. Weight: 737g
order code 128926 — € 449.- £ 392.-

Neumann TLM 102
Large diaphragm condenser microphone
Cardioid pattern, 20Hz-20kHz, 50Ω impedance, 12dB-A equivalent noise level, 144dB max SPL for THD 0.5%, includes SG2 stand mount swivel. Weight: 260g.
Finish: Nickel
o. code 237768 — € 533.- £ 465.-
Finish: Black
o. code 237769

AKG C414 XLII
Large diaphragm condenser mic
9 polar response patterns (cardioid, wide cardioid, eight, hypercardioid, omni and 4 mix pattern), 20Hz - 20kHz, 6/12/18dB pad, 40/80/160Hz HP filter, 140dB SPL max., CK12 capsule, high boost above 4kHz, peak hold LED, lock modus to save preferences, transformerless FET XLR out, incl. CK12 mic head, H85 shockmount, popfilter PF 80, windsreen W414X and alu case.
o. code 240602 — € 885.- £ 773.-

Brauner Phantom Classic
Solid state microphone
Cardioid polar pattern, pressure gradient transducer, equivalent noise <11dBA, signal to noise: 83dB, 20Hz-22kHz, 142dB max SPL @ 0.3% THD, requires 48V phantom power. Includes flexible suspension mount, aluminium case and Vovox cable.
order code 139461 — € 985.- £ 860.-

Neumann U87 Ai Set
The studio microphone classic
Variable large diaphragm microphone, pressure-gradient transducer with double membrane capsule, 3 directional characteristics (omni, cardioid, figure-8), switchable low frequency roll-off, switchable 10 dB pre-attenuation, 20Hz-20kHz, 200Ω. Finish: Nickel. **Bundle includes EA87 shockmount.**
order code 169705 — € 2435.- £ 2126.-

ART Tube MP
Tube microphone preamp
Hand selected 12AX7A valve, limiter, 48V phantom power, phase reverse, XLR and jack I/O, perfect for hard disc recording or as a valve DI.
order code 191529 — € 37.- £ 32.-

Focusrite OctoPre MkII
8-channel mic preamp with A/D
Integrated 8-channel 24bit/96kHz digital output, -10dB pads, 5-LED input metering and direct output on each channel, internal clocking and external via BNC, 8 inputs (2 mic/line/ instrument combo XLR, 6 mic/line combo XLR), 8 outputss jack, 2 ADAT outputs (dual LightPipe), sync-to-wordclock input, JetPLL Jitter-elimination, 19", 1U, weight: 3,6 kg.
order code 236376 — € 389.- £ 340.-

SPL GoldMike 9844
2-channel valve preamp
Discrete Class A solid state, 48V phantom power, phase reverse, pad function, flair presence enhancement, very clear and warm sound. DImensions: 19"/2U
order code 123370 — € 425.- £ 371.-

Universal Audio 710 Twin-Fifty Bundle
1-channel mic/line/DI preamp
Dual path 310V class A valve & transimpedance solid state preamp, +70dB gain, 48V pha3ntom power, -15dB pad, phase invert, 75Hz low cut filter. Bundle includes UAD-2 Solo PCIe DSP card, VST and AU support, LiveTrack™ low-latency live monitoring/recording mode, Mix Essentials II (1176SE Limiting Amplifier Emulation, RealVerb Pro, CS-1 Channel Strip, Pultec EQP-1A EQ Emulation).
order code 244200 — € 679.- £ 593.-

Universal Audio LA-610 MKII UAD-2 Duo Bundle
Classic tube recording channel
Based on the legendary Bill Putnam 610 tube mic pre and EQ, Authentic Teletronix LA-2A style T4 opto-compressor, complete vintage channel strip, mic pre with gain and level controls, true bypass, large metering and improved signal output. **Bundle inkl. Universal Audio UAD-2 Duo PCIe DSP Card.**
o. code 252465 — € 1499.- £ 1309.-

Behringer FBQ1502 Ultragraph Pro
2x15-band graphic equalizer
FBQ feedback detection system, servo balanced ins & outs (jack & XLR).
order code 167770 — € 77.- £ 67.-

Behringer FBQ6200 Ultragraph Pro
2x 31-band graphic EQ, FBQ detection system, jack & XLR I/O.
order code 172364 — € 166.- £ 145.-

FMR Audio RNC 1773
Really nice compressor
Stereo compressor with two modes: ‚Normal' for effective compression and ‚Super Nice' for transparent compression. Controls: threshold, ratio, attack, release and gain. Switches: bypass & Super Nice.
order code 179985 — € 169.- £ 148.-

RNLA 7239 Really nice levelling amplifier
10Hz-100kHz 0.5dB @ 0dBu, clip point: +22.5dBu @ 3% THD.
order code 190166 — € 205.- £ 179.-

dbx 266 XL
Stereo compressor/limiter/gate
Patented OverEasy technology (attack & release times), XLR I/O, -10/+4db switch, stereo link, dbx sound.
order code 131808 — € 175.- £ 153.-

dbx 166 XL
2-channel compressor/limiter/gate, stereo or dual mono, OverEasy or hard knee mode, PeakStop limiting.
order code 131035 — € 266.- £ 232.-

Behringer ADA8000
8-channel AD/DA converter
8x mic preamps with phantom power, 24bit AD/DA, 44.1 & 48kHz, Wordclock or ADAT sync, ADAT I/O, ADAT in can be routed to line outs, mic and line in/outs are routed to ADAT out, excellent expansion for DDX3216 or any interface/mixer with ADAT I/O.
order code 164573 — € 179.- £ 156.-

Benchmark DAC1 Pre
USB audio interface and D/A converter
24bit/192kHz with internal analogue preamp and USB interface, dynamic range: 116dB @ 52kHz, 6x stereo inputs (1x analogue, 3x coaxial, 1x optical and 1x USB), balanced XLR output, unbalanced RCA output, connection for monitors and headphones, UltraLock jitter elimination, internal power supply.
order code 215623 — € 1390.- £ 1213.-

tc Helicon Harmony-G
Compact vocal processor
Harmony arrangement algorithm listens to guitar and voice to follow your music, 6 reverb/delay combinations for vocal and guitar in, 10 presets each with A/B location, selectable harmony voicing, XLR mic in with 48V phantom power, stereo line out (balanced XLR), guitar in and pass-thru for seperate amp/tuner, includes external power supply.
order code 205731 — € 149.- £ 130.-

Boss VE-20
Vocal Performer
Vocal effects dynamic compressor / de-esser, enhancer, delay, reverb, 2 harmonist and pitch correction, LCD display, looper mono, 38 sec., sounds: 30 preset + 50 user, requires optional power supply Boss PSA-230 (order code 102842, not included).
order code 235812 — € 185.- £ 162.-

Digitech Vocalist Live 3G
Vocal harmonizer
Automatic intelligent 3-part harmony, your voice plus 2 more, gender controls, 5 harmony patches, real-time chromatic pitch correction, warmth, compressor, 2-band EQ, low cut, noise gate, reverb, 5 delay settings, chorus and reverb effects, built-in tuner, XLR mic/line in, 48V phantom power, headphone output, incl. power supply.
order code 243163 — € 269.- £ 235.-

tc Helicon VoiceLive Touch
Vocal harmony effect processor & looper
Effects by VoiceLive 2 incl. HardTune and more, 8 voices of NaturalPlay harmonies, MIDI or USB/aux input, 200 presets, 25 favorites, voicing buttons offer instant harmony arranging, VLOOP performance vocal looper, talk button bypasses all effects, XLR Mic In, Guitar in and thru, aux input, MIDI in, USB, headphone out, TRS and XLR output, incl. external power supply unit.
order code 250437 — € 473.- £ 413.-

tc electronic Voice Live 2
Vocal multi FX pedal
Up to 8 voices or 4 doubled harmonies, reverb, tap delay, megaphone and distortion, compressor, de-esser and EQ, MIDI or mp3, LCD display, XLR, line in, aux in, headphone output, coax S/PDIF I/O, backup and audio streaming, incl. power supply. Dimensions: 35 x 7.6 x 21.6cm. Weight: 2.3kg.
order code 225557 — € 699.- £ 610.-

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- 30-day money back guarantee
- free shipping to the UK and Ireland on orders above £159

AKG K 240
Studio stereo headphones
Half open, 55Ω, 88dB/mW, 15Hz-20kHz, ear-enclosing, cable connection, distortion-free playback (new XXL cap), involving sound from new Varimotion system, ideal for Walkman and sound cards. Weight: 240g
~~€ 89.-~~ £ 78.-
order code 153257

AKG K 141 MKII
Dynamic, half open, 55Ω, 18Hz - 24kHz, 200mW. Weight: 225g
~~€ 92.-~~ £ 80.-
order code 206955

Sennheiser HD25 BE
Dynamic DJ headphones
Impedance 70Ω, SPL max. 120dB, frequency response: 16Hz - 22kHz (-3dB), closed system. Includes 150cm cable with 3,5mm mini jack and adaptor für 6,3mm jack. Weight: 140g.
~~€ 145.-~~ £ 127.-
order code 249403

Sennheiser HD 212 pro
32Ω, 12Hz-19kHz.
~~€ 38.-~~ £ 33.-
order code 161045

AKG K271 MKII
Closed dynamic studio headphones
Circumaural design, 55Ω, 200mW max input, 16Hz-28kHz, 104dB/V sensitivity, self-adjusting headband and auto-shut-off feature, 3m cable. Includes 5m coiled cable, 1 pair velvet pads and gold plated mini-jack to ¼" screw-on adapter. Weight: 240g
~~€ 145.-~~ £ 127.-
order code 206951

5m extension cable
¼" stereo jack.
~~€ 6.50~~ £ 5.70
order code 153216

AKG K 701
High-end reference headphones
Revolutionary flat wire voice coil technology, dynamic, open back, 62Ω impedance, 105dBefficiency, 10Hz-39.8kHz, 200mW max input power, hard gold plated 1/4" jack plug, 3m cable. Weight: 235g (not including cable).
~~€ 199.-~~ £ 174.-
order code 185476

Sennheiser HD-650
Hifi headphones
Open back, dynamic, impedance 300Ω, frequency response 10Hz to 39.5kHz, 103dB SPL, 3m cable, 3.5mm stereo mini jack with 1/4" adapter. Weight: 260g (excluding cable).
~~€ 266.-~~ £ 232.-
order code 165585

ESI nEar04 Classic
Active 4" studio monitors
Magnetically shielded, bi-amped, 20W bass + 20W treble, balanced/unbalanced ¼" jack inputs. Dimensions: 20.6 x 13.6 x 14.9cm. **Pair price!**
~~€ 91.-~~ £ 79.-
order code 232799

KRK RP5 Rokit G2
Active studio monitor
45W, 5"+1" speakers, 53Hz-20kHz, magnetically shielded. Dimensions: 27.6 x 18.5 x 25cm. Weight: 8kg. **Unit price!**
~~€ 144.-~~ £ 126.-
order code 213024

KRK RP8 Rokit G2
8"+1", 140W.
~~€ 266.-~~ £ 232.-
order code 213090

ESI nEar05
Active 5" studio monitors
Magnetically shielded, 33Hz-22kHz, bi-amped, 42W bass + 33W treble, balanced XLR & ¼" jack inputs. Dimensions: 25 x 16.6 x 20cm. **Pair price!**
~~€ 169.-~~ £ 148.-
order code 160180

ESI nEar08 Classic
Active 8" studio bi-amp monitors
2-way bookshelf monitors, magnetically shielded, 70W LF & 70W HF amplifier power, 40Hz-24kHz, 1x balanced XLR input, 1x balanced/unbalanced jack input. Dimensions: 35.8 x 25.5 x 32.2cm (HxWxD). Weight: 10kg each. **Pair price!**
~~€ 215.-~~ £ 188.-
order code 222139

Tannoy Precision 6
Passive 3-way studio monitors
6" dual concentric driver, frequency response: 62Hz - 51kHz, crossover frequency: 2,5kHz, nominal impedance: 6Ω, recommended amp power: 60 - 120W @ 8Ω, dimensions: 35,6 x 22 x 26,1cm, weight: 8,9kg. **Pair price!**
~~€ 245.-~~ £ 214.-
o. code 113056

Tannoy Precision 8
8", 54Hz - 51kHz, 44 x 27,2 x 28,7cm, 12,7kg. **Pair price!**
~~€ 283.-~~ £ 247.-
order code 184262

Genelec 8020BPM
Active 2-way studio monitor
Power switch, 66Hz-20kHz (±2.5dB), 105dB SPL peak @ 1m, 95dB SPL @ 1m, 105mm (4") high efficiency bass driver, 19mm (3/4") metal dome tweeter with Directivity Control Waveguide, 3kHz crossover, magnetically shielded, aluminium cabinet, includes wall bracket. Power per channel: 20W (bass), 20W (treble). Dimensions: 22.6 x 15.1 x 14.2mm. Weight: 3.7kg. **Unit price!**
~~€ 279.-~~ £ 244.-
o. code 235038

Yamaha HS50M Stativ-Set
Active 2-way studio monitor
70W bi-amped bass reflex system, 5" woofer, 3/4" tweeter, 55Hz-20kHz, XLR & TRS ins, adjustable input level, mid EQ, room control, high trim, switchable low cut, fully magnetically shielded. **Bundle with 2x HS50M and 1 pair Millenium BS-500 adjustable nearfield monitor stands.**
~~€ 311.-~~ £ 272.-
order code 244787

Adam A7 Special Edition
Active nearfield studio monitor
Finish: black piano lacquer, 6,5" Rohacell / kevlar sandwich woofer, A.R.T. tweeter, frequency range 46Hz to 35kHz, 100W (sin), 150W (rms), XLR and RCA input, dimensions: 18 x 33 x 28cm, weight: 8kg. **Unit price.**
~~€ 377.-~~ £ 329.-
order code 241073

Adam A7 Special Edition
Active nearfield studio monitor
Finish: white piano lacquer, 6,5" Rohacell / kevlar sandwich woofer, A.R.T. tweeter, frequency range 46Hz to 35kHz, 100W (sin), 150W (rms), XLR and RCA input, dimensions: 18 x 33 x 28cm, weight: 8kg. **Unit price.**
~~€ 377.-~~ £ 329.-
order code 244797

Adam A7X
2-way active bassreflex nearfield monitor
7" carbon/rohacell/glass woofer with 100W, X-A.R.T. tweeter with 35W, shelving filter tweeter: >5kHz ~ ±6dB, shelving filter woofer: <300Hz ~ ±6dB, frequency range: 42Hz - 50kHz, crossover frequency: 2,5kHz, SPL max. / pair 1m: 114dB, gain: +14dB / - 8dB, gain tweeter: ±4dB, THD 90dB/1m > 100Hz: <0,5%, inputs XLR and RCA, input impedance: 30kΩ, dimensions: 20,1 x 33,7 x 28cm, weight: 9,3kg. Unit price.
~~€ 429.-~~ £ 375.-
o. code 245513

Genelec 8030APM
Active 2-way studio monitor
58Hz-20kHz free field frequency response, 108dB SPL peak @ 1 m, 130mm (5") bass driver, 19mm (3/4") metal dome tweeter with DCW (Directivity Control Waveguide), 3kHz crossover frequency. Amplifier power: 40W bass + 40W treble, magnetically shielded, aluminium cabinet, balanced XLR input. Dimensions: 28.5 x 18.9 x 17.8cm. Weight: 5.6kg
~~€ 469.-~~ £ 409.-
order code 171718

Yamaha HS80M stand bundle
Active 2-way studio monitor package
120W bi-amped bass reflex speaker, 8" woofer, 1" tweeter, 42Hz-20kHz, XLR & jack input, mid EQ, room control, high trim, switchable low cut, magnetically shielded. **Bundle includes 2x HS80M and 1 pair of Millenium BS-500 nearfield stands.**
~~€ 469.-~~ £ 409.-
o. code 227555

Tannoy Precision 6D
Active 3-way studio monitors
Activ assist, 6" dual concentric driver, frequency response: 59Hz - 51kHz, max. SPL 116dB, crossover frequency: 2,5kHz, amplifier output power: LF - 75W, HF - 35W, XLR/jack combo input, S/PDIF coax input with sample rate from 44,1 to 96kHz, S/PDIF slave out, dimensions: 35,6 x 22 x 37,8cm, weight: 13kg. **Pair price!**
~~€ 473.-~~ £ 413.-
order code 184257

Mackie HR824 MK2
Active 2-way studio nearfield monitor
8.75" low-distortion LF woofer, 1" titanium dome ferrofluid cooled tweeter, twin FR series amps (150W LF & 100W HF), passive radiator provides tight, articulate bass extension down to 35Hz, balanced XLR/TRS and unbalanced RCA inputs. **Unit price!**
~~€ 519.-~~ £ 453.-
order code 138446

Adam A8X
2-way active bassreflex nearfield monitor
9" carbon/rohacell/glass woofer with 150W, X-A.R.T. tweeter with 50W, shelving filter tweeter: >5kHz ~ ±6dB, shelving filter woofer: <300Hz ~ ±6dB, frequency range: 38Hz - 50kHz, crossover frequency: 2,3kHz, SPL max. / pair 1m: 120dB, gain: +14dB / - infinite dB, gain tweeter: ±4dB, THD 90dB/1m > 100Hz: <0,5%, inputs XLR and RCA, input impedance: 30kΩ, dimensions: 25.5 x 40 x 32cm, weight: 12.8kg. Unit price.
~~€ 569.-~~ £ 497.-
o. code 245573

Behringer DJX 750
5-channel DJ mixer
24bit digital effects, BPM counter with time and sync display, Ultraglide faders, XPQ stereo surround effects, 3-band kill EQ, 3-way kill switch, monitor function with master/cue balance control and split option, auto talkover.
~~€ 158.-~~ £ 138.-
order code 223751

Behringer DDM 4000
32bit digital DJ mixer
Beat-synchronized sampler, 4 multi-FX sections, 2 patented BPM counters, digital crossfader and MIDI, 4x phono/line stereo channels. 2x mic inputs with gain, EQ, talk function and FX. 4x stereo channels with gain, programmable parametric 3-band EQ with kill function, fader curve control and flexible crossfader.
~~€ 299.-~~ £ 261.-
order code 206918

Pioneer DJM 400
Professional DJ mixer
2 channels, mic/aux switch (to select 2x mic inputs or 1x stereo line input), fader start, crossfader curve adjustment, talkover (-20dB), peak level meter.
~~€ 433.-~~ £ 378.-
order code 191081

Allen & Heath XONE:42
4-channel DJ mixer
2 turntable ins (RCA phono RIAA), 2 line ins, X-FX send, 3-band EQ, 6dB total-kill EQ on all channels, internal switch-mode PSU, 2 headphone outs (1/4" and 3.5mm), VCA crossfader with curve control, VCF filter with independent I/O, resonance & frequency controls, USB 1.1 port (stereo I/O), XLR mic in with 2-band EQ and level control, finish: black.
~~€ 777.-~~ £ 678.-
o. code 231622

Pioneer DJM-700
Professional 4-channel DJ club mixer
Crossfader assignment, fader start, 3-band EQ (-26dB to +6dB), talk over (-20dB), peak level meter, rotary potis for master output. Dimensions: 32 x 38.1 x 10.8cm (WxHxD). Weight: 7.5kg
Finish: Black
~~€ 811.-~~ £ 708.-
order code 118996
Finish: Silver
~~€ 811.-~~ £ 708.-
order code 119965

Numark NDX 800
Single CD/MP3/USB player
with USB and MIDI interface, beat-synced DSP effects with wet-dry fader, fast, reliable, slot-loading drive, 7" scratch wheel, adjustable start and stop time for vinyl-like deck performance, automatic BPM analyzer and tap tempo for manual BPM entry, pitch ±6, 12, 25 and 100%, bending over pitch buttons, fader start, outputs: RCA and digital S/PDIF.
~~€ 339.-~~ £ 296.-
order code 247251

Denon DN-S 1200
Single CD player
Touch sensitive jog disc, MIDI interface controller (PC/Mac), USB audio, hot starts & seamless looping, 3 platter effects, 4-way BPM counter, memo function, pitch range & deep pitch resolution, vibrant tube display with 2-line character text support, loop A/B trim, power-on-play, slot-in CD player, D-Link, 4 built-in DSP effects, next track function with cross fader, dimensions: 21.5 x 8.7 x 23.2cm.
~~€ 449.-~~ £ 392.-
order code 205617

Pioneer CDJ-350
Single CD player
Plays MP3, AAC, WAV, AIFF, USB (flash recorder, disc drives), USB A/B connection, includes Recordbox music management software, beat display function for better scratch and looping performance, auto beat loop function, dimensions: 21.8 x 29.6 x 10.8cm, weight: 2.7kg, system requirements: Windows VISTA/XP, Mac OSX 10.4 or higher.
~~€ 589.-~~ £ 514.-
order code 249908

Pioneer CDJ-850
Single DJ CD player
MP3, AAC, WAV, AIFF, 24bit/48kHz, HID and MIDI control by USB, new audio output system with Wolfson DAC processor, quantized beat loop, advanced auto beat loop, plays Audio CD, CD-R, CD-RW, USB, frequency response: 4Hz - 20kHz, dimensions: 30,5 x 36,4 x 10,5cm, weight: 3,3kg, incl. Rekordbox music database management software, Mac OSX (10.4.6 and higher), Win 7 & XP.
~~€ 885.-~~ £ 773.-
order code 251872

Pioneer CDJ-2000
Professional multiformat single CD player
Plays Audio CD, CD-R, CD-RW, USB, SD, DVD-R, DVD-RW, USB sources and SD cards, includes Rekordbox software, quantized beat loop, compatible with MP3, AAC, WAV, AIFF and Tonium Pacemaker data, 24bit/48kHz soundcard, HID and MIDI controllable from USB, frequency range: 4Hz - 20kHz, dim.: 32 x 40,6 x 10,7cm, weight 3.9kg.
~~€ 1775.-~~ £ 1550.-
order code 238585

Native Instruments Traktor Pro
DJ software
With new optimized interface, automatic error correction, 4 decks for up to 4 tracks or loops, more than 20 effects, for PC and Mac.
~~€ 95.-~~ £ 83.-
order code 219788

Native Instruments Traktor Scratch Duo
Professional DJ system
Control digital music files using turntables or CD decks, up to 2 decks simultaneously, controllable by CD player and MIDI controller, USB 2.0 audio interface, bus powered, 4x high-gain outputs, 4x inputs with phono preamps, 8x LEDs, 2x vinyls and 2x CDs.
~~€ 169.-~~ £ 148.-
order code 227391

Native Instruments Traktor Scratch
Professional DJ system
Enables control of digital music files using turntables or CD decks, 24bit/96kHz low-latency USB 2.0 audio interface Audio 8DJ with Cirrus Logic AD/DA converters, MIDI I/O, 8x hi-gain outs, 8x ins, 2x optional phono preamps with high impedance, mic in, 2x timecode records & 2x high resolution CDs, DJ software based on Traktor 3, for Mac OSX 10.4.8 and XP+ SP2.
~~€ 289.-~~ £ 252.-
order code 111859

Hercules DJ Console RMX
DJ controller
Integrated USB audio interface, 2 jog wheels, 6 faders, DJ buttons for DJing control, FX and loop buttons, mic in, headphone out, 4x line outs (¼" mono balanced jack), 4x RCA outs, 2x stereo ins, includes VirtualDJ DJC edition software (PC/Mac).
~~€ 269.-~~ £ 235.-
order code 208113

Numark Mixdeck
DJ workstation
With CD/MP3/USB decks, mixer, computer, MIDI interface, dock for iPod, mixer with EQ/rotary kills, effects with beat sync (echo, filter, flanger, pan, phaser), fader start, looping, sampling, tempo: 6, 12, 25 and 100%, controllable over MIDI software without timecode over USB, 2 displays, Mac OS X or Windows XP/Vista, incl. power supply and USB cable.
~~€ 859.-~~ £ 750.-
order code 245624

Project SAM Symphobia 2

Sample Library

Symphobia's sequel probes deeper into the psyche of the sampled orchestra.

DAVE STEWART

The success of Project SAM's Symphobia is founded on a simple fact: many users lack the time, patience or know-how to painstakingly build sampled orchestral arrangements one instrument at a time, and so appreciate a library that features full, ready-to-play strings, brass and woodwind family sections as well as ensembles of different instruments playing together. Add to that Symphobia's impressive array of hits, rips, clusters and glissandi and its large soundscapes section, and the net result is a powerful, impressionistic and easy-to-use orchestral collection that was well received across the board.

Building on the principles established in the 2008 original, SAM have created a follow-up, entitled (no doubt after much deliberation) Symphobia 2. Contrary to rumour, it does not incorporate the original library (though it was recorded in the same concert hall): it's a new, separate, much larger, supplementary collection, which takes over where its predecessor left off. (You can read the original *SOS* review of Symphobia at www.soundonsound.com/sos/dec08/articles/sampleshop.htm)

Symphobia 2 is formatted for Native Instruments' Kontakt 4.1 sampler, which runs stand-alone and as a plug-in on Windows and Mac machines. Kontakt Player 4 software (which doesn't allow editing) is included on the installation DVDs — but if you already own the full version of Kontakt 4.1, you won't need this player (in fact, if the installation software detects the full version of the program residing on your system, it won't install it anyway!). Thanks to Kontakt's new lossless compression method, the 33GB of 24-bit sample data occupies only 18.2GB of disk space when installed.

Phobile Tones

Since the 'all together now' instrumental approach of the original went down so well, Symphobia 2 offers a lot more material in the same vein. The producers go straight for the jugular with a powerful 'full orchestrator' patch containing strings, brass and woodwinds playing together. With this, you can keyswitch between splendidly forceful and meaty staccatos, looped sustains and a set of long notes in which higher instruments double the bass notes across four consecutive octaves. An optional, additional grandiose timp-roll layer makes these octave samples ideal for the final, triumphant note of your latest orchestral masterpiece. Then again, if you'd rather end on a big fat tutti major or minor chord, there are plenty of them in the library too.

Much as I enjoyed the lively staccatos of the combined strings and woodwinds patch, I initially found the woodwind element hard to detect — because the loud, emphatic strings bow attack (a Symphobia trademark) tends to mask it. Further listening revealed a bass clarinet in the low register and a trace of an oboe higher up. Though the wind instruments' contribution appears subliminal, you'll notice a marked difference in tone if you compare this sound to a strings-only

Project SAM
Symphobia 2 **£979**

PROS
- Great, strongly played orchestral ensembles, solo instruments and effects.
- Features pre-orchestrated mixed-instrument sections and tutti samples not found elsewhere.
- Real interval-based legatos.

CONS
- Not cheap.

SUMMARY
Like its predecessor, Symphobia 2 takes some of the slog out of creating sampled orchestrations. Full, ready-to-play sections provide instant colour and drama, the musical effects are great and the sheer force of the loud deliveries enables them to hold their own against the heaviest backing. Though there are gaps in its instrumentation and articulations, this is a top-quality, highly effective and versatile collection that is guaranteed to liven up your scores.

patch. On a less subtle note, you won't have any difficulty hearing the bright, perky xylophone the producers recorded playing along with violins and woodwinds, an inspired touch that has created a unique and colourful orchestral timbre.

A new set of full string section articulations complements those in the original Symphobia: these include energetic and bouncy short spiccatos optimised for film action scenes, trills and a set of beautifully played, emotive and precisely co-ordinated crescendo-diminuendo swells. Unison sustains from the original Symphobia are also included, but there are no full string-section pizzicatos. I greatly enjoyed the 'grace note performer' patch played by unison violins and violas: urgent, emphatic short staccatos that mutate into semitone grace notes when played loud, supported in the bass by massively powerful tutti staccato stabs. This creates instant orchestral drama — and I could jam for hours with this sound. The octave cellos and basses are also very good, and absolutely nail the threatening, low-strings grumble that says to audiences 'something bad is about to happen'.

Instrumentation

FULL ORCHESTRA
- Strings, brass & woodwinds (oct).

MIXED ENSEMBLES
- Strings & woodwinds.
- Violins & flutes (leg).
- Muted violins & flutes.
- Violins, woodwinds & xylophone.
- Violins, violas & French horns (leg).
- Cellos & solo bassoon (leg).

STRINGS
- Full string section.
- First & second violins (oct, leg).
- Violins (oct).*
- Violins & violas.
- Violas.**
- Cellos.
- Cellos & basses (oct, leg).
- Brass
- Three trumpets (oct).
- Eight French horns.
- French horns & trombones (leg).
- Solo French horn (leg).

WOODWIND
- Two flutes & solo piccolo (oct).
- Two flutes & solo clarinet (leg).
- Solo flute (leg).
- Bass clarinet & contrabassoon (oct, leg).

MISCELLANEOUS WINDS
- Irish low whistle (leg).
- Uilleann pipes (leg).

MISCELLANEOUS NOISES
- Orchestra tuning.
- Concert hall noise.

Instruments are played in unison unless stated otherwise.
(oct) = played in octaves. (leg) = includes legato sustains. * Marcato short notes only.
** Ponticello tremolo only.

SAM built their reputation on orchestral brass samples, and the selection here maintains their usual high standard: eight French horns performing sustains whose fierce attack can be attenuated with the mod wheel, powerful staccatos, semitone trills (reminiscent of a swarm of aggrieved giant bees) and some real-life, deliberately »

Alternatives

You'll find excellent individual strings, brass and woodwind ensembles in **Vienna Symphonic Library** and **EastWest Quantum Leap Symphonic Orchestra** — however, you won't hear them all playing together in various combinations, as they do in Symphobia 2. Its musical effects and processed soundscapes are also unrivalled by any other orchestral collection, which means that the only real alternative to Symphobia 2 is the original Symphobia.

» discordant pitch-bends, in which notes start off in unison and gradually drift apart. The higher brass is represented by three trumpets playing long notes in clean, well-tuned octaves. Though this is by no means a comprehensive brass collection, its strong, commanding tones are fit to summon an emperor.

Legato Obligato

I was pleased to see that in this library SAM have, for the first time, implemented interval-based legato sampling for some instruments, recording all up-and-down chromatic scale steps up to an octave. Pioneered by Vienna Symphonic Library in 2003, this technique has become increasingly common in orchestral libraries, but I believe this is the first time it's been applied to mixed ensembles.

The results sound excellent, and the instrumental pairings chosen for the legatos are delectable: two flutes and a clarinet sound sweet, smooth and lyrical, while the oily, villainous low-end octaves of a bass clarinet and contrabassoon duo positively scream 'baddie!' I liked the melancholy sound of cellos and solo bassoon, though once again the woodwind element is a little overwhelmed by the strings. When it comes to orchestrating John Williams-esque, uplifting, melodic motifs, you've a choice of unison French horns and trombones, or a superb violins, violas and horns patch. In a more intimate vein, I found both the solo flute and solo

Project SAM's interval-based legato sampling for mixed ensembles is an industry first.

For speed of access, Symphobia 2 presents all articulations played by each instrument or section in one keyswitchable patch.

French horn legatos to be agreeably liquid-sounding, and the folksy slides and gracings of the Irish low whistle and Uilleann pipes are also very playable, as long as you remember they're only mapped to the notes of the D-major scale!

SAM advise that the legatos (which are one dynamic only) are designed for melody lines at tempos of up to 150 bpm (playing quarter notes). I found that they can handle faster tempos and, in the case of the solo winds, will do a passable imitation of trills and grace notes if you turn off the release trails. Though monophonic by nature, these legato instruments can be rendered polyphonic and non-legato via a keyswitch. A matching, keyswitchable staccato articulation is also provided for each one — a very sensible and practical idea. Hats (or any other item of clothing you'd care to nominate) off to SAM for this welcome development. (By the way, though most of Symphobia 2's instruments can be edited in Kontakt 4, the legato patches and sample pool are locked for editing.)

Disturbia

His Fabness Sir Beatle Paul, referring to the improvised ascending orchestral cacophony in 'A Day In The Life', said, "The strings were like sheep — they all looked at each other: 'Are you going up?'" In the Symphobia series, the string players are not sheep — they're wolves. In the 'wild rips' section, they lead the rest of the orchestra in a series of uproarious, cataclysmic swoops before turning round and devouring the woodwind section. The brass players sound equally psychotic, contributing nasty, low-pitched snarls over scary, atonal chords and dissonant crescendos that drip with menace. 'Symphobian Stacks' sees multiple musical effects layered over orchestral bass drums, piatti cymbals and a low octave of synth-bass swells to devastating effect; one particularly intense sonic event starts off sounding like a distant fly-by of Alpha Centaurian attack craft before crossfading with a long, massive, detuned, subsonic gong-and-orchestra crescendo rumble. Do not play these samples to young children.

It's not all horror. In the 'dreamy textures' section, strings, horns and woodwinds play pleasant, pastoral phrases in a selection of easy-on-the-ear harmonic settings, including a classic 'let's go back to your childhood' whole-tone-based mini-fantasia. As in the original Symphobia, the musical effects and phrases are supplemented by an enjoyable set of processed 'Dystopia' soundscapes

In Symphobia 2's 'Dystopia' section, the mod wheel can be used to apply extreme filter settings to the processed soundscapes.

System Requirements & Activation

- PC: Pentium or Athlon 1.4GHZ processor with Windows XP (SP2)/Vista/Windows 7.
- Mac: Intel Core Duo 1.66GHz processor with OS 10.5 or higher.
- Both platforms: 1GB of RAM, 20GB of free disk space.

After installation, the library can be activated by entering its serial number at Native Instruments' Service Centre web page. It's also advisable (though not obligatory) to register it with Project SAM, in order to take advantage of any free bonus content they might offer in future, and/or access technical support, should you need it.

comprising low-pitched, evolving drones, electronic alien warblings, glitchy electronic splutterings and a fine, brain-juddering selection of violent impacts, explosions and piano demolition noises. Lurking within this mayhem is a small collection of suspended cymbals and taiko-like ethnic drums, and a lovingly-recorded chain saw. (Every composer should have one.)

Kontakt Mics

Like all SAM libraries, Symphobia 2 presents its samples in a choice of mic positions, in this case 'stage' and 'close'. Both contain traces of the concert hall reverb; when you load an instrument, both mikings are immediately available, selectable by a switch on the front panel. The Kontakt GUI gives instant access to important parameters such as EQ, ADSR envelope settings, and an effects menu that includes convolution reverb, delay, stereo spreader, compression and a resonant filter. Added to the extensive keyswitching facilities, such a wealth of features renders the distinction between sound library and virtual instrument almost entirely academic.

Conclusion

The only minor criticisms I have of this library are firstly that some instrument ranges seem somewhat curtailed, and secondly that the liberal mixing and matching of unisons and octaves within tutti patches occasionally caused me problems. For example, I found it disconcerting that the high register of the 'full orchestrator' unisons suddenly sprouts an upper octave of trumpets when played loud. The saving grace is that the programming is always designed to make the sound bigger and more dramatic, which is, after all, what most media composers want.

Symphobia 2 doesn't set out to cover every instrument and articulation under the sun, and thus would not be a good choice for recreating all the pointillistic detail and subtle gestures of a full orchestral score. What you get instead is orchestral samples with attitude, presented in instantly playable sections that can be a great inspiration for composing. It's safe to say that most users will find something they really enjoy in this large collection — and if absolutely nothing in it stirs you, it might be advisable to consider an alternative career in accountancy.

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Waves Kramer HLS & PIE

EQ & Dynamics Plug-ins For Mac & PC

SAM INGLIS

Eddie Kramer is a producer and engineer who knows a thing or two about rock & roll, so when he gives his name to plug-ins modelled after his favourite hardware, we expect great things...

It's been a couple of years now since Waves introduced the first of their Signature Series plug-in bundles, modelled after the actual hardware used by some of the best-known engineers in the business. The plug-ins in the series fall into two categories. On the one hand, there are 'task based' plug-ins dedicated to particular instruments such as bass or electric guitar, which model the entire processing chain that an engineer like Tony Maserati or Chris Lord-Alge would typically use on those instruments at the mix, and present the results in a friendly interface with just a few controls. However, Waves have also produced straightforward, faithful emulations of individual pieces of vintage hardware from those engineers' studios.

So, in addition to the Eddie Kramer Artist Signature Collection, which contains 'task based' plug-ins reproducing Kramer's favoured vocal, bass, drums and guitar processing chains, Waves have now introduced the two more conventional processors under review here. Kramer HLS reproduces an input channel from the famous Helios console found in the Rolling Stones Mobile, while Kramer PIE is an attempt to recreate the vintage Pye compressor in software.

A Fresh Start

These are the first Version 7 Waves plug-ins I've tried, and the installation procedure has changed since I last tested any. If your studio computer is connected to the Internet, you can run an 'online' installer which downloads and installs only the plug-ins you're after. If not, you have to download and run the entire 1GB-plus Waves 7 complete installer, which is a bit of a nuisance. It also turns out that Version 7 plug-ins cannot co-exist in a system with any other Waves plug-ins earlier than Version 6, so when you run the Version 7 installer, it automatically uninstalls any Version 3, 4 or 5 Waves plug-ins on your system — even if you only want to demo a single Version 7 plug-in. There are plenty of people out there still using these earlier bundles, myself included, and this is a major disincentive to investigate the new plug-ins.

On the plus side, at least Waves are now using the standard iLok authorisation procedure whereby you redeem a licence code at your iLok.com account, then download the licence in the usual way.

Helios Is Other People's

The Helios equaliser has been modelled before by other manufacturers, and an official, licensed version using the Helios name is available for Universal Audio's UAD2 platform. Waves' version doesn't use the Helios name, but the look and feel leaves you in little doubt as to what it's supposed to be. It also goes further than the UA version, in that it models the characteristics of the line and mic preamp stages as well as the EQ itself. (Waves are keen to point out, too, the attention to detail that has gone into the ballistics of their virtual VU meter, which does indeed bounce in a comfortingly musical fashion.)

The equaliser is a quirky design, which takes some getting used to. Simplest of the three bands is the high shelving filter, which is operated by a gain knob stepped in 4dB increments. The middle band offers a choice of eight centre frequencies, with a freely variable knob that applies either boost or

Waves Kramer HLS & PIE £338

PROS

- Helios-style EQ sounds great, and including the preamp stage makes a big difference.
- PIE compressor has a unique character that is sometimes perfect.

CONS

- Waves Version 7 installer automatically uninstalls Version 5 plug-ins and earlier, even if you're only demoing a new plug-in.
- So authentic you might need one of Waves' noise-reduction tools too...

SUMMARY

These are two vintage emulations that have rock & roll character in spades.

Test Spec

- Dell Inspiron laptop with 2GHz CPU and 4GB RAM, running Windows XP SP3.

attenuation, depending on the setting of the Pk/Tr switch. The low band is particularly odd. Moving the main knob clockwise from zero brings in a low shelving filter, this time in 3dB increments. Moving it the other way selects one of four low/low-mid centre frequencies that can be boosted (but not cut) using the adjacent dial.

A few minutes' trial should be enough to convince anyone that this is not a surgical equaliser. The high and low shelving bands allegedly turn over at 10kHz and 50Hz respectively, but their effects are obvious well into the mid-range frequencies, while the mid-range band is about as coloured as EQ gets. And as if that weren't enough, you can ramp up the gain on the preamp stage to add further saturation. Thankfully, doing so doesn't actually affect the level of your signal, it merely replicates the side-effects. As these can include a large dose of hiss, Waves have added a 'low noise' option.

Without access to the real thing, I can't tell you whether Waves' is a more accurate emulation than Universal Audio's, but I was surprised by how different the two are. I duplicated an audio track and inserted the Waves on one copy and the UA on the other, with polarity reversed. There was some cancellation, but the signal level dropped only by a few dB.

I had thought the UA Helios a really nice EQ, and still do, but the Waves version struck me as noticeably superior: it has a hairy, larger-than-life quality that makes the UA version sound polite by comparison. I also preferred Waves' user interface, which is larger and more friendly, and allows you to set knob positions by clicking on the numbers as well as by 'turning' the knob itself. The only irritation concerns the factory presets, which all seem designed to sound impressive by making the signal louder.

I liked the Kramer HLS plug-in in many roles. The high shelf adds a nice gloss to drum overheads, while the mid-range boost brings electric guitars right into the listener's face, and adds a chewy bite to rock vocals. The preamp emulation is hit-or-miss; I didn't like it on drums at all, but on anything with strings, it sounds wonderful. I found plenty of places to use this EQ in my mixes, but for me, the killer application is on acoustic guitar. Add 8dB of high shelving EQ, engage the low shelf, crank the line preamp up as high as it will go, and you get a glorious, rich sheen without a hint of harshness. Lovely — albeit still obviously hissy, even in low-noise mode.

Nice As PIE

As sought-after British vintage gear goes, the classic Pye compressor is up there with the Helios EQ, though the only other plug-in emulation I'm aware of is the one in the Focusrite Liquid Mix, which I haven't used. Again, Waves' version doesn't use the Pye name, but in other respects it's visually close to the original. The control set is simple: there's a fixed attack time, a choice of six release times and five compression ratios, plus Threshold and Output gain controls to set how hard the compressor is driven.

Waves say that the original is renowned for being "among the best bus compressors of all time" — not that you'd know this from their preset library — so my first thought was to try the Kramer PIE plug-in on the master bus. Perhaps it was just me, but I didn't really get on with it in that role: with the fast release settings I normally use, its action was a bit too obvious, and there was too much pumping going on for my tastes. It can be a little disconcerting to hear obvious compression artifacts even when the gain-reduction needle is barely moving!

Elsewhere, by contrast, its unsubtle nature is exactly what is wanted, and if, by "bus compressor", Waves mean something you might strap across a drum group to add unbridled rock & roll excitement, I'm with them 100 percent. It's likewise perfect for smashing room mics and, in more moderate doses, works nicely on acoustic guitars and vocals. On bass guitar, meanwhile, its fixed attack time makes note attacks 'pop' in quite a distinctive way, which wasn't to my taste but may well be to that of others. Though it's not the most versatile processor in the world, this is one compressor that's not shy about making its presence felt, with a sound that is both distinctive and powerful.

HLS+PIE=OMG?

Like many faithful emulations of vintage units, these two plug-ins are limited to doing what the originals can, and Waves have made no attempt to add extra functionality beyond giving you the option to reduce noise and hum. As a result, they have pretty restrictive parameter sets, and it would be limiting by modern standards if they were your only EQ and compressor.

That said, within half an hour of installing Kramer HLS, it was all over the mix I was doing at the time. I've never heard another plug-in EQ that is quite so nice on acoustic guitars, and it's no slouch on other rock instrumentation either. It might not always be the best thing for the job, but it nearly always makes a positive contribution. By contrast, the PIE compressor fairly often struck me as just being the wrong processor for a particular role — but when it's right, it's very right indeed, particularly on a drum submix. I think anyone who demos these plug-ins will be tempted, so it's just a shame that doing so is such a pain for anyone running earlier Waves collections on their studio machine. ■■■

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Neve Manufacturing Australia CUSTOM SERIES 75

Analogue Mixing Console

Thought you'd seen the death of the large-format console? Think again, as you read our exclusive preview of this modern take on classic Neve designs.

HUGH ROBJOHNS

The introduction of any new analogue console is always greeted with considerable interest, but one that purports a Neve heritage doubly so. The launch of the Custom Series 75 at the Sixth SAE Alumni Convention in Berlin in October stirred up a hornets' nest of debate across the Internet, both before and after I had an exclusive preview of what's currently the only prototype, with its designer Bruce McBean.

Overview

The first thing to say is that this console has been developed and manufactured entirely independently of the British AMS Neve company, and it appears that it will be manufactured and supported exclusively by a company set up in Australia by Tom Misner, founder of SAE. AMS Neve have agreed that the Custom Series 75 console can be marketed with the phrase 'Powered by Neve' — which essentially refers to the fact that the new console's circuitry is derived from the original Neve 1081 input and 1272 line-output modules.

The prototype I saw, which was two years in the making, will receive a few further modifications before the production units leave the factory, and the control software is still being fine-tuned. Most of these tweaks, though, are very minor — like the detail of the stand construction, the layout of part of the rear connector panel and some extra control-label legends.

The console is constructed from extremely strong 'bins', each holding eight individual 40mm-wide 'N2081' channel strips, together with a 10-channel-wide master section. The channel and master bins can be placed in any order. The 32-channel prototype is surprisingly compact: although it can mix 80 inputs to the stereo output if required, every control can be reached quite easily from a central, seated position. Configurations of up to 64 channels are planned, but for where there are more than 32 channels a larger (by 80mm) centre section will be required, to accommodate the additional mix-bus and routing-relay hardware.

The estimated cost of a 32-channel model including 2245 compressors is around $64,000 plus shipping, with moving fader automation and HUI control being made available as cost options at some point in the future. That may not be cheap, exactly, but if it sounds like the modules it is based on, that's actually quite competitively priced!

Modern Classics

This console is essentially a hybrid of Neve designs from the 1970s and some more up-to-date elements. The channel mic-preamp and EQ circuitry, for example, is derived from the classic Neve 1081 module, but it uses discrete surface-mount transistors. In fact, this is probably the first discrete transistorised console to be made in years!

Misner says that this style of construction will make future servicing more efficient — for example, through no-quibble exchange of individual channels.

The discrete circuitry is based around Neve BA338 gain stages and BA283 output stages. However, it must be said that while the circuitry is similar, it isn't identical to the original designs: Bruce's aim was to recreate the original function and sound, but not the form, because to do that would have proved way too expensive.

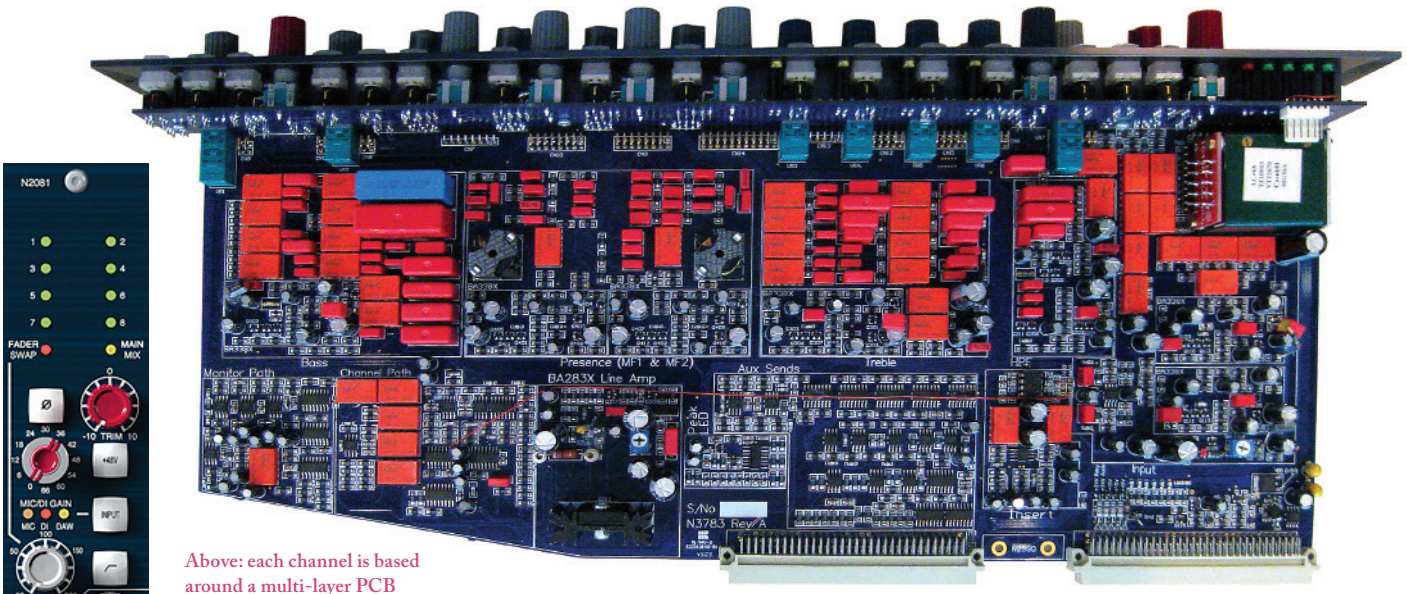
Interestingly, as Bruce and his team used 'SPICE' (Simulation Program with Integrated Circuit Emphasis) modelling to optimise the original circuit designs, they discovered some deficiencies in the originals along the way. For example, a weakness was found in the high-pass filter circuit so, having addressed this weakness in the new design, the Series 75 design runs 10dB more level through the channel path — and offers better dynamic range as a result.

Gain-stage biasing arrangements have also been modified, to improve temperature stability and reduce distortion significantly. However, most of the 'Neve' sound comes from the output stage and the transformer, and in the Series 75 the 1081's Class-AB output has been replaced with the Class-A stage from the 1073/1272 modules, complete with an MJE3055 transistor feeding the traditional LO1166 gapped-core transformer. The desk has a lot of these chunky transformers, and they're mounted in the chassis rather than on the channel strips.

The 1081 channel path is essentially the same as on the original, but with a single 10468 transformer in the channel strip that handles the mic, DI, and attenuated line and DAW channel inputs (the line and DAW inputs bypass the first 45dB gain stage too). Other cost-reducing tweaks include omitting the rarely-used low-pass filter, and giving the high-pass filter a continuous rotary control, rather than a switch. The horribly expensive triple-layer input-gain switch has also been replaced with a much simpler switch, controlling relays to give gain ranging from 0-66dB in 6dB steps (instead of the original's 5dB increments) plus a variable ±10dB trim for a maximum gain of 76dB (the original's was 80dB). This re-configuration is claimed to give marginally better performance at a much lower cost.

Similar modifications have been used in the EQ, with relay switching for the high and low bands, and separate frequency and gain controls instead of dual-concentrics. The 'off' positions provide a genuine section bypass too. Each channel has three mono and one stereo aux sends (the latter can be routed to two stereo aux buses), with pre-post and channel-monitor path selection via a push-button that's integrated into the level control. By default, the channel insert point is post-EQ, but jumpers on the PCB allow it to be set pre-EQ if preferred.

With the channel insert point bypassed, the entire signal path comprises discrete transistors. However, if the insert is engaged the signal is dispatched and received via high-spec op-amps: a Texas DRV135 line driver and a National LM4562 to handle the return. In fact, LM4562s are used in all the critical stages, such as the current-summing mix-bus amps and the monitor-path gain stages. Although the eight groups don't have insert points as such, the group direct outputs can be used as sends »



Above: each channel is based around a multi-layer PCB and fitted with around 1300 components. Left: the channel strip is laid out as you'd expect, given the designs on which it is based.

» and the corresponding DAW inputs as returns.

All the switches, relays and connectors handling audio signals are gold-plated, while capacitors in the signal path are either polypropylene film or Rubycon ZLH-series electrolytics, both chosen for their sonic performance and long life. The single-sided nature of the transistor circuitry inevitably means that a lot of electrolytic capacitors are required.

Retro *Or* Modern

Most modern consoles use current-summing (virtual earth) technology, but the vintage Neve designs employed a voltage summing arrangement using transformers — a design in which the bus impedance and channel output level is affected significantly by the number of source channels connected.

The Series 75 console, however, has both types of bus available. The eight groups and the 'retro' stereo mix bus use voltage summing (with transformers in and out). Unlike Neve consoles of the '70s, though, the voltage mix buses are fully balanced and very short, with channel routing performed via relays in the master section; the buses themselves are only 150mm long. Unusually, the channel pan control is after the channel output transformer, and the bus impedance seen by the channel and mix amp remain constant, regardless of the status of the channel routing relays. This avoids the loading losses that classic Neve consoles suffer as a channel is routed to more buses.

The second (non-retro) stereo mix bus, the monitor mix bus and the AFL buses use current summing, so the user has a choice of mix topologies, and they can be selected globally or on a per-channel basis. The two stereo mix buses (voltage and current) are subsequently combined, along with the group outputs and the monitor stereo mix bus, onto another local current mix bus, before being passed to the master fader. The stereo mix signal is then routed via optional and patchable 2254 compressors and an insert point to the master stereo outputs.

Extending the sonic options even further, the master outputs are also provided with dual topologies: one set of stereo master outputs follows the old-school 1272/transformer route, while the second set is electronically balanced using LM4562 op-amps.

This unusual flexibility means that the user can choose an all-retro signal path (fully transistorised with four transformers between input and stereo output — or six transformers if you route via the groups!), or an all modern op-amp path (with only the input transformer in the way — and even that can be avoided if the monitor return path is used). With a little thought, it's also possible to use a hybrid path comprising both technologies, with the option of tracking in retro mode and mixing in modern mode, for example, or vice versa.

Compressor & Patching

The 2245 compressor design is basically the same as the original, but the diode bridge that unit employed is no longer available, so Bruce uses matched diodes instead, which, it turns out, produce less asymmetric distortion and a subtly smoother sound character. The input and output of the two 2245 modules, if installed, can also be accessed externally if the compression is not required on the stereo bus. A calibration oscillator and pink-noise source are also included, as are a studio loudspeaker (SLS) output and built-in talkback and studio-listen facilities.

Although a patchbay will be available as an option, since each channel has separate line and DAW inputs and the monitor path can also select between Line and Monitor inputs, in many cases sources can be hard-wired to the console without compromising flexibility. Most of the rear-panel interfacing is via DB25 connectors using the usual Tascam wiring standard. Input-channel bins have eight XLR and quarter-inch sockets for the mic and DI inputs respectively, plus five DB25s to handle the eight line and DAW inputs, the direct outputs and the insert send and returns.

Master I/O & Power

The master section has XLRs for the main speakers, two-track monitoring input and stereo mix output, while everything else is on DB25 connectors, including a spare socket »

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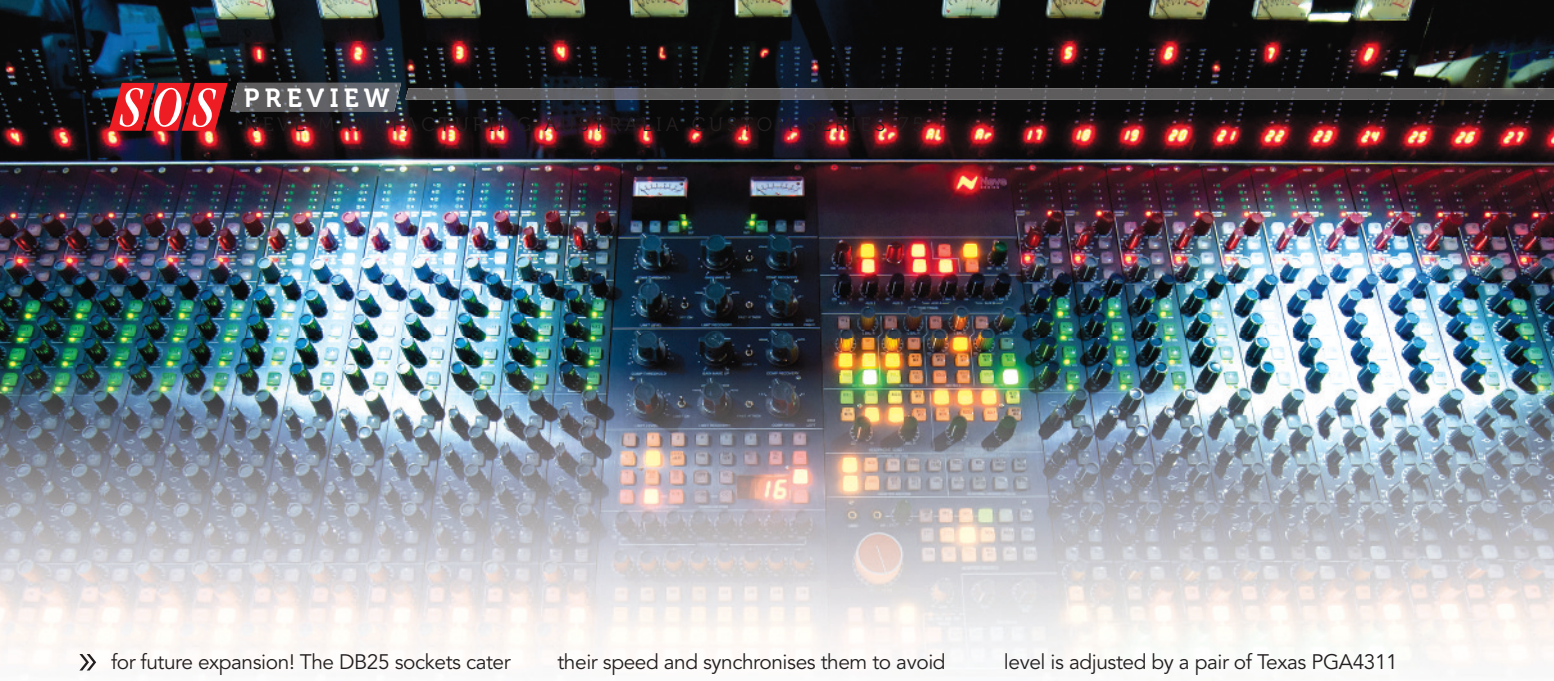


» for future expansion! The DB25 sockets cater for 7.1 inputs, Alt2 surround outputs, four stereo reverb returns, stereo monitoring speakers, aux sends, red-light logic, Studio LS and talkback in/out, headphones, mix inserts and two-track inputs, mix outputs and 2245 I/O, group outs/sends, stem outs, and group/tape returns.

There's also an Ethernet socket, the PSU input and provision for power to the HUI DAW control option — because motorised faders require ±12V power, which the normal PSU doesn't provide. In fact, five regulated voltage rails are supplied from the main PSU, each regulated again in the meter bridge of each bin, and again on the individual channel-strip PCBs. The audio relays operate on a 24V supply, while a 27V rail is dropped to 24V for the transistorised audio circuitry. Dual ±15V rails power the op-amp circuitry and a 9V rail is dropped to 5V for all the logic. Finally, of course, there's a 48V phantom supply. A pair of ultra-quiet cooling fans is located in the central section, and the central microprocessor monitors their speed and synchronises them to avoid 'beating'. Fusable resistors are included on each channel-strip PCB board to provide protection, too.

As I mentioned earlier, fader automation will be available as an option, although the hardware and software has yet to be developed. A basic level of DAW control is also planned, probably using the HUI protocol, with 32 channels of fader and cut automation.

Monitoring & Metering

The monitoring section caters for three sets of stereo speakers fed from 12 sources, and the huge volume knob is scaled, amusingly to 11½. (*Spinal Tap* has a lot to answer for!) The Alt 2 output can also be reconfigured for 7.1 surround monitoring, complete with user-adjustable channel solo, cut and level trims using the group status buttons. The stereo monitoring outputs all use relay attenuators, with facilities to dial in level offsets between the different outputs. When Alt 2 is switched to 7.1 mode, however, the level is adjusted by a pair of Texas PGA4311 four-channel volume controller chips. A large LED display normally indicates the selected channel, but while the monitoring level is being adjusted it displays the output level. Some clever mono monitoring switching options permit muting the left or right speaker, or the left or right source, as well as listening to the L/R difference signal.

PFL, AFL, solo-in-place and solo-in-front modes are all included, and a nice feature is that the channel and monitor paths can be isolated from the solo bus by pressing their respective pan controls. In a similar way, the channel and monitor faders can be swapped by pressing the channel HPF control. The PFL signal can also be routed just to the 3.5mm headphone socket on the monitor panel, or to the studio loudspeaker output, if required. There are actually three headphone amps built into the console, using the National LME49710 output driver op-amps feeding the top-panel 3.5mm output, plus two rear-panel quarter-inch outputs — both of which have additional line-level outputs to drive external power amps or distribution systems.

The channel and monitor-solo and cut buttons can be configured for momentary, latching or auto-switch operation, and there's even an option to configure the action of the buttons so that they operate on the press or the release! Four stereo effect returns can be routed independently to the two headphone outputs, mix or monitor buses, each with level and balance controls, plus solo and solo-isolate buttons.

Each channel is equipped with a bar-graph meter, and a further eight assignable bar-graph meters sit above the master section. Eight small VU meters monitor the group outputs, and a further two show the main outputs (with an associated phase meter). The channel bar-graphs can be switched to show channel, monitor or direct output levels, or the fader position (for fader automation), and

Bruce McBean, the Custom Series 75's designer.

switched between VU and PPM modes. The master-section meters can be switched four ways to show main, monitor, control room and AFL outputs; or the eight groups; or the aux outputs and PFL levels; or the four stereo-return levels.

Every channel bin is controlled via its own microprocessor, which communicates with the central section on a dedicated RS485 balanced data bus with a centrally distributed data clock. This microprocessor control of all switch functions is useful when the desk is constructed, as it allows almost fully automated testing — which reduces the manufacturing costs significantly — but, more importantly, it allows instant recall of all desk switch configurations (there are currently eight instantly recallable scenes).

The production console will have a web server built in for configuration, testing and other utilities, accessed via the Ethernet port. A pair of BNC connectors on the top of the meter bridge provide power for optional gooseneck lamps.

Channel PCBs

The channel printed circuit cards all have either four or six layers, with extensive ground planes to minimise crosstalk, and the data control signals are kept well away from the audio. There are about 1300 components on the boards, but clear labelling aids fault-finding and reveals the careful thought that has been given to enable easy maintenance — such as leaving clearance so that the potentiometers and switches can be removed easily, and the incorporation of additional 'via' connections, so that if the board is damaged when replacing a worn-out potentiometer, the wiring can be reconnected using these reserve connections. These are very thoughtful considerations that reveal Bruce's years of experience in servicing consoles and other equipment.

The Custom Series 75 comes with a one-year labour and five-year parts guarantee, but it is clearly designed to remain serviceable for 20-plus years. Similar attention to detail can be seen in the fact that all the buttons are engraved, rather than printed, and the panel legends are laser etched — so the legends won't ever rub off.

The Custom Series 75 is an impressive console to look at. All the small control knobs on the channels are made by the same British company that makes the larger winged knobs used on the optional 2245 bus compressor modules, and these bespoke knobs work well ergonomically. They also convey a very attractive vintage feel, which, combined with the ultra-modern illuminated push-buttons, makes a very impressive sight. The design and build quality is extremely high, yet with a careful eye on cost to ensure that the console remains competitive alongside the likes of API's 1608 and SSL's AWS900.

Verdict

While it's obviously not the same as a 1970s Neve console stuffed with 1081s, the Custom Series 75 follows the same design topologies and shares the same core sound character, but it also offers an ultra-clean, modern signal path for those who require it. That the DAW-control features (both current and planned) are limited to the HUI protocol may concern some potential users, but as a straight analogue console, this is a very attractive beast in every sense, and it should do well.

£ A 32-channel console including compressors will cost in the region of $64,000 plus shipping. Flying faders and HUI control will be available as cost options.
T Neve Manufacturing Australia.
E Email via web site.
W www.customseries75.com

Black Lion PM8
Passive Summing Mixer

The PM8 is an eight-channel, passive summing mixer, although a 25-pin D-sub connector on the rear panel allows a second unit to be coupled, providing 16 mix channels in total using a female-female parallel cable. The eight main balanced inputs are connected via 'combi' sockets on the rear panel, accepting TRS jacks or XLRs, and each input has its own 10k:10k Edcor transformer — which accounts for a large part of the surprising weight of this unit.

Each input channel has its own rotary pan and rotary fader control on the front panel, plus an associated mute button that disconnects the input from the passive stereo mix bus altogether. The output from the passive mixing networks is split three ways. One feed goes direct to a pair of rear-panel XLRs labelled as passive outputs, and this feed is unbalanced and at low level, so an external preamp would typically be required to restore the signal level, but this does give the option of choosing the gain-stage sound character. The other two splits are both amplified by solid-state gain stages to make up for the losses of the passive mixers. One set of outputs is dispatched via balancing transformers (nominal 600Ω types), while the other is presented by dual active output buffers, all connecting through two more pairs of rear-panel XLRs. The master output levels are controlled by two stereo gain knobs. Master 1 controls the output level of the passive and transformerless outputs, while Master 2 controls the level of the transformer-coupled output.

The PM8 is constructed in a similar way to the AM/CHA1 equaliser (reviewed on page 30 of this issue). It uses the same compact linear power-supply module, which is switchable for 120/240VAC mains supplies, and has the same worrying chassis-grounding problem (although the company have undertaken to ground the metal case directly in production units). It also suffers the same haphazardly mounted front-panel power LED, and employs the same surface-mount gain-stage components and the same Edcor input and output transformers.

In use, the mixer does exactly what it says on the box, with the transformers undoubtedly adding a distinct vintage character, which fills out the bass in a very analogue way. Having the option to use an external gain stage, or to take the restored level output via active buffers or another transformer certainly provides plenty of sonic flexibility. The absence of a centre detent on the pan controls is a little disappointing, as is the lack of a unity gain mark on the master controls — but these issues are easily worked around.

Hugh Robjohns

£ $895 plus shipping.
W www.blacklionaudio.com

Seymour Duncan STK-S10 YJM Fury
Hum-cancelling Guitar Pickups

Stacked humbucker pickups designed to replace standard single-coil pickups are nothing new, but most simply strive to deliver a single-coil tone while offering the noise-suppression ability of a humbucker — and resistance to interference is very desirable in the studio, where today's low-noise equipment makes guitar-related hum and buzz seem worse than ever.

Seymour Duncan's Fury pickups, conceived in collaboration with guitarist Yngwie Malmsteen, are somewhat different, because they're specifically designed for players who need a hot pickup with plenty of mid-range weight. The result of their endeavours is the STK-S10 YJM Fury three-pickup Strat set, which is available as a conventional set of components, or ready-wired on a pick guard for those who like an easy installation.

I fitted the Furies to a Swamp Ash Strat I was building and tested it both live and in the studio. Given Malmsteen's playing style, you'd expect these pickups to suit shred, hard-rock and metal players — and indeed they do — but I was impressed by their versatility.

As expected, the Furies are noticeably hotter and more weighty in the mid-range than regular Strat pickups but they don't overwhelm the distinctive Strat tone. Certainly, the warmer mid-range and sweetened highs make them very overdrive-friendly, but they also work well played clean. I found the bridge pickup less brash and more solid-sounding than those on my regular Strats, reminding me somewhat of a P90. Furthermore, the middle pickup, which many players use only in combination with the neck or bridge pickups, delivers a very useful tone in its own right, again combining power with great definition. I was also surprised at the 'in-between' tone gained from combining the middle pickup with either the bridge or neck pickup, as the sound can thin out quite a lot when using single-coil pickups. Here, the familiar 'phasey' tone was underpinned by a healthy lower-mid weight. All three pickups respond well to playing dynamics, allowing you to 'dig in' for more aggression.

Overall, these pickups turned a typical Strat into something very versatile. Using amp and desk EQ, you can get very close to a classic Strat tone, but you can also make the bridge pickup lean towards a P90-equipped SG sound. If you want to cover most of the tonal bases using a single guitar that also has great noise immunity, this set has a lot to offer. A one-trick pony it is definitely not. *Paul White*

£ £87.95 each; Three-pickup set £261.95; loaded pickguard £357.95. Prices include VAT.
W www.ariauk.com
W www.seymourduncan.com

T-Rex Fuel Tank Chameleon

Guitar Pedal Power Supply

As anyone who's ever tried to put together a pedalboard incorporating a few vintage pieces, a digital/modelling pedal and a modern boutique item or two will know, the days of the one-size-fits-all power supply are long over. Even the comparatively modest pedalboard that I use throws up a requirement for 9VDC, 9VAC, 12VAC and 18VDC at the same time. Manufacturers of dedicated pedalboard power supplies have clearly noticed this trend, and T-Rex have been adding ever more options to their multi-output supplies. The Danish pedal makers' Fuel Tank Classic model, their five-way, isolated Fuel Tank Jr and their high-current Juicy Lucy models are now joined by the highly versatile Fuel Tank Chameleon.

The Chameleon model offers five isolated, centre-negative 2.1mm outputs: three are switchable between 9VDC and 12VDC; another is switchable between 9VDC and 18VDC; and the final option uses either output 5 or 6 (they can't be used simultaneously), which offer 9VDC, 12VDC or 12VAC, with voltage options set by a miniature DIP switch. Switchable (230/115V) AC voltage is a welcome inclusion, with mains power input via detachable IEC. Total supply capacity is 1.5A, with a nominal 300mA limit on each output. As all the outputs are isolated, capacity is not transferable — so you can't use 600mA on one output just because you aren't using one of the others, but you could use a paralleled, 'current-doubling' cable (not supplied in the otherwise exemplary cable set) across two identical outputs. Isolation also confers greater immunity to hum and other noise problems, and prevents a fault on one circuit from pulling down the others.

In action, the Fuel Tank Chameleon does exactly what you'd want it to do. Assembling all my 'awkward stuff' to run outputs 1, 2, 3, 4 and 6 simultaneously, with a daisy chain on output 1, produced nothing but clean power, with no power-supply-related noise issues at all. The device runs hot, but not excessively so, and is a safe weight to mount via Velcro. A second 18VDC output would be great for Fulltone fans like me, as would a 24V output for users of some Electro-Harmonix pedals, but the designers had to draw the line somewhere, and the choices made are perfectly logical for the target market. This one certainly gets the *SOS* stamp of approval.

Dave Lockwood

£ **£155 including VAT.**
W **www.t-rex-effects.com**

Electro Harmonix Freeze

Sound Retainer Effects Pedal

Electro Harmonix's Freeze pedal performs much the same job as the freeze function on the Roland VG99 guitar synth: trigger it during the decay portion of a guitar chord and it will loop the sound to form an indefinite, smooth-sounding drone, with an almost organ-like quality. Although it fits into a 'nano' pedal format, it consumes too much current for battery operation to be practical, so it comes with a 9V DC power supply. Other than the input and output jacks, the box has just a single knob, a three-way toggle switch and a footswitch, with a red LED to indicate when the freeze function is operating. The dry sound always passes through the unit at its normal level, and the knob controls the amount of 'frozen' sound added to it — so once you have a drone going, you can play over it normally.

The toggle switch provides three modes of operation: Fast, Slow and Latch. In Fast mode, pressing and holding the pedal produces the frozen sound, which keeps running until you release the footswitch, at which point it stops immediately. Slow mode works in a similar way, but allows the drone to fade in a little more slowly, and to fade out again when the footswitch is released. Three different rates of 'slow' are available — the slowest fade time being just over three seconds — and you select them via a simple power-on ritual. I found the middle setting, with its 200ms attack and one-second fade, the most comfortable to use.

Latch mode works much like the Fast mode, but you can take your foot off the pedal and the sound will continue to sustain. When you next press the switch, it freezes whatever you've just played — which is useful for creating continuous pads over chord changes. To bypass this mode, you simply tap the footswitch twice. There's no hard bypass, but to my ears there's very little sonic difference between having the pedal in circuit and not.

Providing you can keep your foot in sync with your playing, this pedal is simple to use and produces a very usable sound, with no giveaway end-of-loop glitches. For the smoothest results, wait a second after playing a chord to hit the switch. The resulting sound can then be treated using the usual array of modulation and delay effects, to add movement and interest.

My only gripes are that there's no way to kill the dry sound — which could be useful for producing drone-only sounds — and no way to separate the dry and freeze signals to separate outputs, so that the freeze signal feeds one amp and the dry sound another.

In the studio, the Freeze pedal provides a practical means for guitar players to produce good-sounding pad parts, for treatment with rotary simulators to make them more organ-like, or with chorus for pseudo-strings. Freeze may be a bit of a one-trick box, but it produces a novel effect, and one that's not easy to achieve by other means.

Paul White

£ **£89 including VAT.**
W **www.ehx.com**

One To Another

Transferring Projects Between Different DAWs

Exporting a project from one DAW to another can be frustrating — but there are ways and means...

MATT HOUGHTON

We're often asked how to transfer projects from one software DAW to another. It might not be an everyday need, but sooner or later most of us will wish we could do exactly this, with the minimum of pain and inconvenience — so let's explore the benefits, limitations and idiosyncrasies of the protocols and tools that aim to help you.

Why Aren't DAW Files Interchangeable?

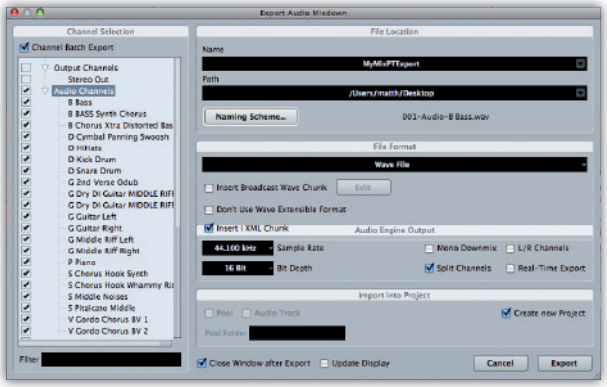
DAWs do pretty much the same job, but they all work slightly differently: they may use different plug-in protocols (TDM and RTAS plug-ins for Pro Tools, Audio Units, VST or even Direct X for others); they may offer different amounts of gain above unity; and the implementation of audio and MIDI routing, of automation, VCA grouping, or the way they handle multi-output virtual instruments or crossfades may also differ.

Even with different versions of the same DAW 'family', you may find that an 'LE' or 'lite' version can't open a project created in the 'full-fat' product, simply because the full version includes functionality that's missing or disabled in others. Newer versions of a DAW may include additional functionality, and different plug-ins from previous versions, as plug-ins have been updated, or licensing deals with third-party suppliers of older plug-ins expire.

That said, there are several areas of commonality, so it's always possible to transfer at least some data: all use a timeline, and offer multiple mono or stereo audio tracks; they support plug-in effects, processors and instruments; they generate automation data, probably using MIDI, to control effects and virtual instruments; the job of summing signals together on a bus is a simple mathematical process... and so on.

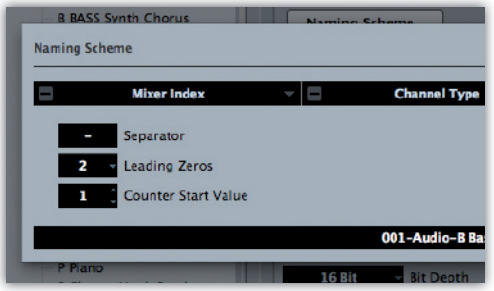
Media Savvy

Before exporting a project, consider what media you plan to use for the transfer. Assuming you're not just switching between DAWs on the same machine, it's easy and inexpensive to use an external hard drive, and even USB pen drives are now typically large enough to hold a full project (just don't try to run the project directly from it!).



If your DAW allows you to bounce all tracks simultaneously, it can save a lot of time and nervous energy!

By default, Macs and PCs use different drive formats (NTFS on Windows; HFS+ on a Mac) and without additional software, neither OS can write to the other's drives. To get around this, you have two options, the first of which is to use a FAT32-formatted drive (which is read/writable on both operating systems), and the second to install software (freebies are available) that enables your OS to read drives of the other format. If you're working with very large files, such as HD video, or audio projects in a single ZIP file, you'll need to use the latter approach, as FAT32 can't deal with single files over 4GB in size. If you're planning to transfer via the web instead, none of this matters, though there are some further considerations that I'll discuss later.



When bouncing tracks down as audio, try to make use of options for organising the naming. Adding an incremental number at the front means that your tracks will be imported to a new DAW in the correct order.

MIDI

Let's start by looking at MIDI file transfers, because that's the easiest element of any project to export. MIDI files are read and written in the same way by every DAW, and by a good many hardware machines too. So if you have several tracks of programmed drum beats, piano and string parts, it's a straightforward affair to save the MIDI file and re-open it in another piece of software.

You'll only be sending the standard MIDI information, of course, such as note on, note off, program changes and controller data; you're not exporting any virtual instruments or audio files themselves.

However, if the same instruments and patches are available in the second DAW, you'll be able to use the MIDI files to get those instruments to play the same things back — although you won't have any effects or level automation on the virtual instrument output channels. It's for this reason that I tend to bounce virtual instruments as audio, which can be edited and processed in the usual way, and only use MIDI as a backup.

There are a few quirks to watch out for, such as identical instruments having different presets on each machine, or how multitrack MIDI files are exploded on to different tracks, but exporting and importing MIDI files is generally a pretty painless process.

Audio

The most basic, and still the most reliable (if not the most flexible) way to transfer audio and virtual instrument tracks is to bounce each down as a continuous audio file, with all tracks starting at the same point (eg. bar 1, beat 1). That way, when you import the files into the other DAW, all tracks will line up as they should. That's the way we usually work at *SOS* with our Mix Rescue projects, for example.

As long as you use a standard, uncompressed audio file type such as WAV, BWF or AIFF, any DAW will be able to read these files — so even if you don't plan to transfer your projects, it's a good practice to archive them in this way after you've completed the project. I prefer to bounce two versions: one pre-fader (pre effects and level automation); and another post-fader, with effects and processors 'printed'. That way, whatever changes in technology come along, you'll be able to re-open the project in any multitrack software.

Depending on which DAW (and version) you use, bouncing the files may appear a daunting process. Previous versions of Cubase, for example, required you to bounce each track separately, which could take an eternity with a large project; and if you're working with analogue gear, or on Pro Tools, you'll be limited to bouncing tracks in real time — which can be frustrating with long projects (a 20-minute radio drama, for example). »

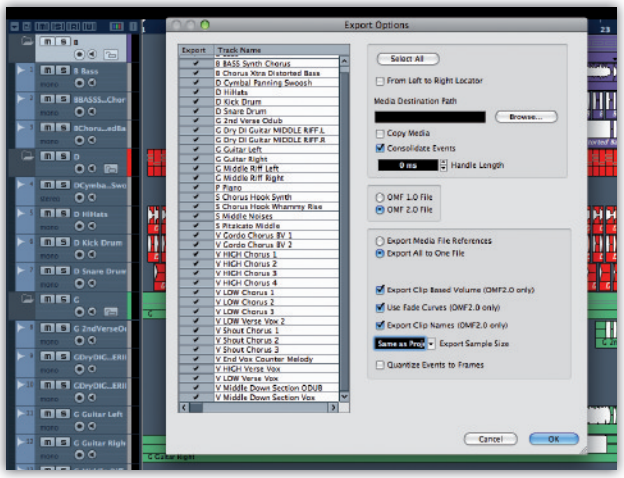
Open Media Framework (OMF)

OMF files save the following information:
- Tracks and track numbers.
- Clip positions.
- Slip edits.
- Fades and crossfades (as destructive edits).

Some important things are discarded:
- Volume & Pan
- Automation
- Effects

OMF files can include the audio as part of the file, or save the media files separately and create an OMF file that references them. If you plan to use OMF, it's wise to experiment before using it on critical projects, so that you can be confident it works, and that you know you've properly prepared your target session — for example, by setting the correct tempo in a Logic session into which you plan to import the OMF file.

Many DAWs can create OMF files, which allow you to transfer audio tracks, with a bit of information about levels, between different pieces of audio and video software.



» Many DAWs also offer some form of 'consolidate' function (as it's called in Pro Tools). The facility varies from DAW to DAW, but it's essentially a means of merging the various clips/regions on each audio track into a single audio file. At its most basic, your edits will be intact, but you won't get the results of any plug-in processing. This is perfect if you've tracked and comped some parts in a studio using Pro Tools, but want to export the session for mixing in another DAW at home.

In some DAWs, the process is a little more fiddly than in Pro Tools. In Cubase, for example, you may need to draw a new region at the beginning of each track, then select all clips in the track and join them with the glue tool, before using the Events To Part command. In Logic, the process is done per track, so you need more user input than in Pro Tools — but on the plus side, it presents several options, including whether to bounce the parts with the effects in place. Remember that consolidation typically only applies to the audio tracks, and not to any send or group channels. So depending on what DAW you're using, and what stage the project has reached, bouncing each track down or consolidating might, or might not, be the right way to go.

Broadcast Wave Files

Another audio-only option is to export time-stamped broadcast wave files (BWAV or BWF), which are 'time-stamped'. Many sequencers are able to automatically line them up at the correct point on their timeline, such that the basic edits and arrangement changes remain intact.

This approach is suitable in the same scenarios as bouncing or consolidating, and the pros and cons are largely the same, although not *all* DAWs will be able to read the time-stamp (most can, but check the target DAW's manual to be sure). There's another benefit, though: you don't end up creating huge, continuous audio files that take up storage space. OK, hard drive real-estate isn't that costly now, but if you're planning to transfer projects via a web-based delivery system (an FTP site, Dropbox or Yousendit, for example), file storage and upload/download speeds are a real issue.

OMF (Open Media Framework)

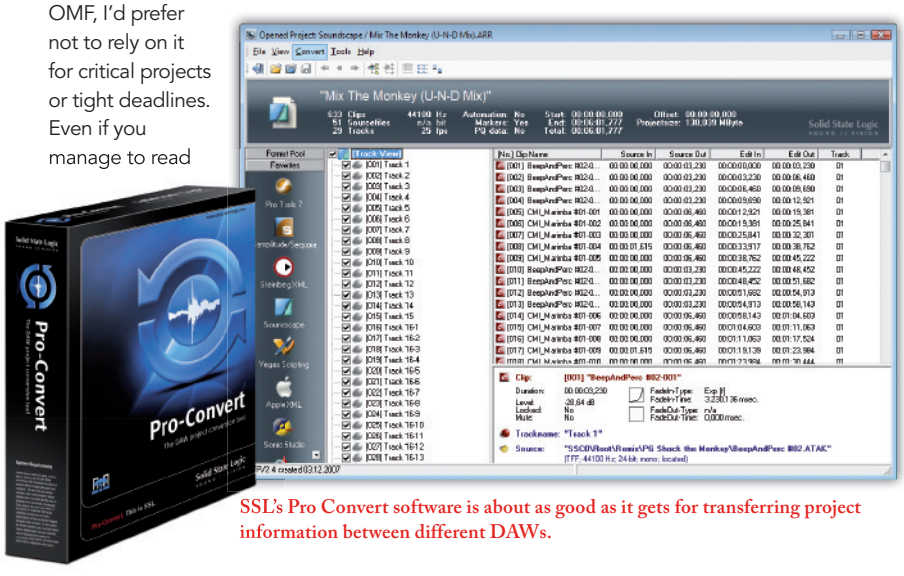
The Open Media Framework (OMF) protocol was developed as a means of transferring audio clips, along with a basic amount of information (see box), between different audio and video software. Most (not all) full versions of the leading DAWs include the option to import and export OMF files, although many 'lite' versions don't. (Be aware that OMF support is a cost option for any versions of Pro Tools below HD.)

OMF is a great idea, but I've found it less great in practice: sometimes it works, sometimes it doesn't, and different DAWs seem to require you to prepare a 'target project' in different ways. I've had different results exporting from Pro Tools to Logic, Logic to Cubase, Cubase to Pro Tools, and so on. So, while you *can* get results from OMF, I'd prefer not to rely on it for critical projects or tight deadlines. Even if you manage to read and write OMF files, there's still no provision for transferring information about level and pan automation, or any information relating to plug-in instruments or effects, or MIDI. So, other than enabling you to bundle everything into a single file, and putting the clips on the right track numbers, it offers little more than time-stamped WAVs.

Pro Convert

OMF is about as far as DAW manufacturers have travelled towards any sort of interoperability standard, but there have been attempts to create more flexible and reliable conversion software, the most successful and best-known of which is Solid State Logic's Pro Convert (a more recent incarnation of Cui Bono Soft's EDL-Convert). It aims to allow you to transfer project information between a vast array of different DAW and NLE (video) packages. There are too many to list here (around 40 at the last count, including the different versions — check www.solidstatelogic.com for details) but it covers the usual suspects such as Logic, Sonar, Pro Tools, Cubase and Nuendo, and Digital Performer, as well as catering for OMF and other standards.

Pro Convert costs about £250 on the street, and is currently a stand-alone application for Mac or PC. It allows you to open DAW projects, refine a few options, and (hopefully) spew them out as a different DAW's project or XML file. So, for example, it can open a Pro Tools PTF file, and convert it to a Cubase XML file. Like OMF, it can only act as an interpreter for a certain amount of information — it still doesn't 'do' MIDI, static fader positions, or plug-ins — but there are several advantages over OMF. It translates level and pan automation, for example, so as long as automation is enabled on



SSL's Pro Convert software is about as good as it gets for transferring project information between different DAWs.

"The most basic — and still the most reliable, if not the most flexible — way to transfer audio and virtual instrument tracks is to bounce each down as a continuous audio file."

the original project's tracks, you'll get the same levels in the target DAW. Also, as the recipient of a file, if a studio sends you the Pro Tools file, you don't have to ask them to re-export it as OMF: you have the power to open and convert the project yourself. Or, if you use Pro Tools, you can open a Logic file with interleaved stereo files and convert it to Pro Tools with split-stereo files.

My own tests threw up similar results to those I've read in reviews: for the most part, it works remarkably well, though there are occasional glitches. There were a few timing issues with the odd clip, and a few issues with warped (like Acidised) audio files in Cubase. But generally, it does the job very well. Unfortunately, Pro Convert doesn't yet run on 64-bit operating systems, though SSL say that the application is most definitely still being developed.

Virtual Instruments & Effects

What Pro Convert does not do is transfer any details about plug-ins — and I'm afraid that this is an area where you'll keep drawing a blank. For the time being, all you can do is to save presets and reload the plug-ins in the new DAW, then reload the presets (don't forget to transfer any samples used by your samplers, too!).

You may think that the holy grail is to be able to transmit all the information about the plug-ins contained within a project. To a certain extent, I'd agree, but even if you *are* able to transfer all this information, you'll still be relying on both DAWs/machines to have the same plug-ins installed. If you're switching between DAW software on your own machine, that's less of a problem than if transferring to someone else's; but even then you won't be able to transfer a DAW's in-built plug-ins (say, a Pro Tools EQ to a Cubase Channel EQ or vice-versa).

Closing The DAW

That, in a nut shell, is it. There is currently no magic-wand to make everything you've done in one DAW readable by another, without committing at least some of it to audio. If you're planning to start collaborating with someone, and have yet to decide which software to work with, it really does make sense to invest in the same setup. That way, you can be sure that you'll be able to read each other's files. I suppose that this is one of the reasons why Pro Tools remains so prevalent in the professional world, too — people just know that they can open projects from other studios if they need to.

Perhaps we'll see the likes of Pro Convert evolve: I'd love to see it incorporated into leading DAWs, just like OMF in the past — and knowing that SSL are still developing this software gets my hopes up — but in the meantime, your options remain limited.

Whichever transfer method you're contemplating, it's probably still a good idea to bounce everything as audio files as a back-up — and as far as I'm concerned, you might as well work with those if they'll do the job for you: at least bouncing things down as audio in this way will force you to commit to edits and mix decisions and get on with things, just as printing to tape used to do! ///

Secrets Of The Mix Engineers: Mike Poole

Thirty years after Led Zeppelin ended, Robert Plant has reached a second career high. His latest hit album was tracked and mixed by Mike Poole, using a mouth-watering selection of vintage equipment.

PAUL TINGEN

The million-dollar question for every ageing rock dinosaur is how to avoid going the way of the dodo, or, at the very least, Rod Stewart. The question has been answered in various ways, in most cases by pigheadedly refusing to adapt to age and changing times, and simply carrying on as if everything is as it ever was. There is another strategy, however, which may be described as growing old gracefully. One of its main proponents has been the now 62-year old Robert Plant.

While it greatly frustrates Led Zeppelin fans, it has done no harm to Plant's credibility that he's steadfastly refused multi-million dollar offers to tour with his former band. In addition, during his post-Zep solo career Plant has repeatedly managed to defy expectations and explore different and unanticipated musical directions. Plant's upward career path started with an album of bluesy rock covers, *Dreamland* (2002), followed by a mixture of rock and world music, *The Mighty Rearranger* (2005). Only moderately successful from a commercial point of view, both albums were critically lauded, and each garnered two Grammy Award nominations.

The roof blew off in 2007 with *Raising Sand*, a duet album with bluegrass singer Alison Krauss. An exploration of Nashville-influenced Americana produced by T-Bone Burnett, the album won a staggering five Grammy Awards, including for Album of the Year, went platinum in both the US and the UK, and sold three million worldwide. Three years later, Plant revived the name of the first band he played in, the Band Of Joy, a '60s psychedelic folk outfit that also featured Zeppelin drummer John Bonham. But rather than a harking back to that long-distant past, Plant's new album, *Band Of Joy*, is a continuation of *Raising Sand*, in that it also explores Americana with Nashville overtones. It again features a well-known female vocalist, Patty Griffin — although in a much less prominent role than Krauss's on *Raising Sand*. There's a change in the production team as well, with Burnett making way for country guitarist, singer-songwriter and producer Buddy Miller, who co-produced *Band Of Joy* with Plant, while the recording and mixing processes were helmed by Mike Poole.

Atmosphere & Character

Band Of Joy had, a month after its September release, reached number three in the UK and five in the US hit parades, and seems destined to achieve similar commercial success to *Raising Sand*. Billed in a press release as "a timeless plunge into authentic Americana", *Band Of Joy* contains country, gospel and soul covers from a few decades ago and before, but also more current songs by Los Lobos ('Angel Dance'),

'Angel Dance'

Written by David Hidalgo and Louie Perez
Produced by Robert Plant and Buddy Miller

Richard Thompson ('House Of Cards'), Townes van Zandt ('Harm's Swift Way'), and two atmospheric epics by the Minnesota 'slowcore' band Low ('Silver Rider' and 'Monkey'). *Band Of Joy* is also more experimental sonically than its predecessor, though both were recorded in Nashville.

Engineer and mixer Mike Poole is a Nashville native who was planning to study commercial photography at Middle Tennessee State University in the early '80s, encountered "one of the first recording programmes in the country" there, and promptly decided that this was his future. He worked in various studios and became independent around 1990. "I went one step up on the autonomy ladder, and took a corresponding pay cut," he laughs. Over the last 25 years, Poole has become a familiar and highly respected face in Nashville's studios, clocking up credits such as John Prine, Martina McBride, Patty Griffin, and many, many more. Poole got the gig for the *Band Of Joy* recordings via Buddy Miller, whom he regularly helps out whenever the guitarist doesn't want to do the engineering himself.

According to Poole, the only pre-production that took place was the song selection and choice of musicians, with the *Band Of Joy* becoming Miller on guitars and vocals, Darrell Scott on mandolin, guitar, accordion, pedal, lap steel and banjo, Byron House on bass, Marco Giovino on drums, and Patty Griffin on vocals. Bekka Bramlett also contributed backing vocals to two songs on the album. Poole: "Buddy and Robert talked a lot about what the album was going to be, but did not prepare many arrangements before we went into the studio. The musicians and I got some discs with songs before we went in, and there were some emails and conversations, so everybody knew what the influences and some of the potential songs were, and beyond that Robert and Buddy only made sure that we all went in with a similar mindset. The arrangements were worked out in the studio, and if they worked, great, and if there was no magic, we simply moved on. The songs were mostly recorded live in the studio, with very few overdubs or edits. Usually, we did at most two or three takes of each song. Most of what we were doing was about creating the atmosphere and character for the songs, and not sweating over details.

"Most of the album was recorded at Woodland Sound, which is the studio of Gillian Welch and Dave Rawlings. It had been a commercial studio until about 10 years ago, when they purchased it. They have since used it as their private facility. We knew them and we wanted a place that was both technically appropriate and private, and they happened not to be working on their own stuff at the time, so they were gracious enough to let us use it. We went in on the first of December 2009, spent a couple of weeks recording, and then came back in at the beginning of February for another four days of tracking and a few overdubs. From there we went straight into mixing in studio A at House of Blues Studios, Nashville. Because of band members' schedules, I had to do some overdubs while I was mixing. I spent two weeks mixing, and the whole album was finished by mid-February, except for one track on which we later altered a vocal and I tidied up a couple of things."

Welch and Rawlings are well-known purveyors of high-quality Americana and lovers of analogue equipment, and it is to be expected that their Woodland Sound Studio is full of the vintage, weird and wonderful. "Yeah," Poole confirms, "the studio is full of lovely old vintage and ribbon mics, Telefunken and Neve preamps, Pultec EQs, Fairchild compressors and so on. They also have an old Neve 16-channel, four-bus broadcast desk, plus an small API sidecar. Buddy and I also brought a lot of our own gear. In the end, I only used their Neve desk for monitoring. Instead, all our input channels were mics going into discrete mic pres, then via compression or EQ if needed, and then straight into a 16-track Studer A800 tape machine."

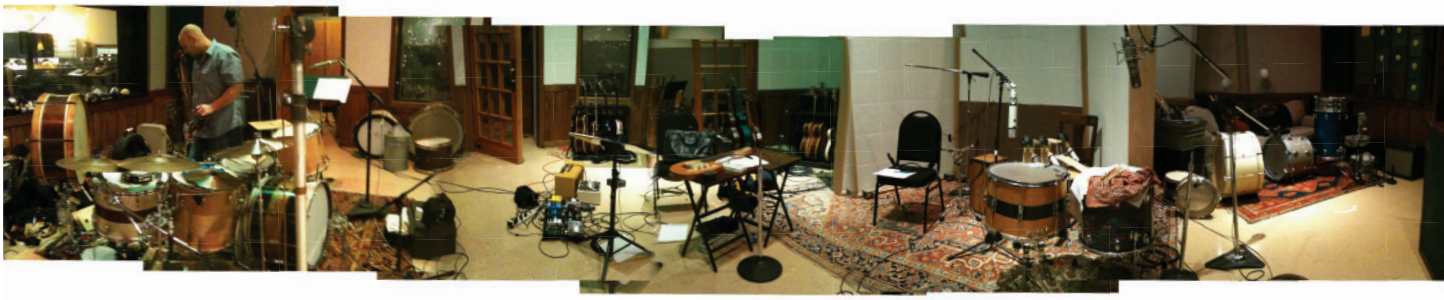
Initial live takes were recorded to 16-track, two-inch tape, before being transferred to Pro Tools for overdubs and mixing.

Limitations

And so, predictably perhaps, Plant, Miller and Poole went for the classic record-live-into-a tape machine approach that's been at the heart of Americana for 60-odd years. The famous American producer Bill Bottrell once opined to this writer that Americana will soon bite the dust, as it's wedded to the analogue recording medium, while Pro Tools is moving music to a much more urban and synthetic place.

"I would agree with Bottrell in terms of the difference between the sonics of analogue and digital," says Poole. "There are genres for which DAWs work wonderfully, but I don't think that they are ideal yet for this kind of music. One of the reasons Buddy and I like working with each other is that we both like going to the same places sonically, so it's not a battle to try to get across what each of us wants, and we knew that the *Band Of Joy* album would be much more suited to analogue from a sonic perspective. One of the arguments of proponents of digital is that the playback sounds exactly like the input. That may be true, but with analogue tape the playback often sounds better than the input. I prefer to go with the medium that makes things sound better!

"Robert also has a preference for working in the analogue domain: it's where he's from, and it sounds familiar and attractive to him. But most of all I think he likes working with the limitations of the medium, the fact that you can't easily slide tracks backwards and forwards in time and you might have to punch in a whole bar »



This panoramic shot shows the live room at Woodland Sound during tracking.

» rather than a single note. Plus there's the vibe and approach that came from limiting ourselves to 16 initial tracks. This meant that we had to make decisions early and we couldn't mess around with laying down many different drum scenarios, or whatever, which you tend to do when you have tons of available tracks. It's easy to get lost in a world of endless options.

"I don't see a Doomsday scenario like Bottrell does, however. For example, DAWs encourage you to work with a constant tempo to facilitate editing, but it is quite easy not to have a click going and to use it as you would a tape recorder. The trick is to remember that and not allow yourself to be tied to the ways in which the software tries to get you to work — don't let the tool overly influence the way you work. And I think that there will always be younger people coming along that will be inspired by the music from the past and that will adapt DAWs to work for them, rather than allow the computerised, linear way of looking at and organising music to become a stumbling block. They will just hit Record and Play and will be able to work beyond what the medium expects them to do."

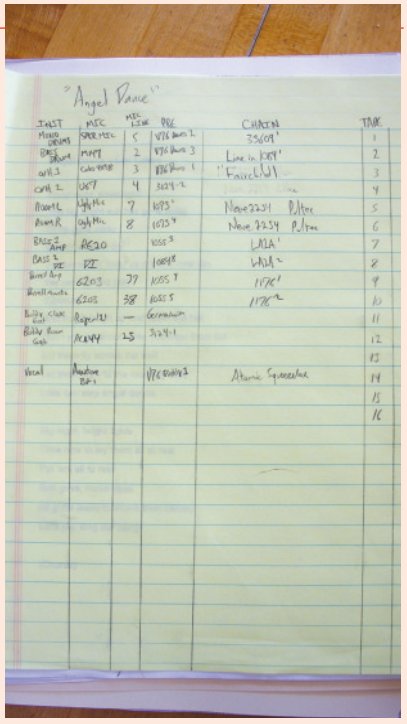
Failsafe

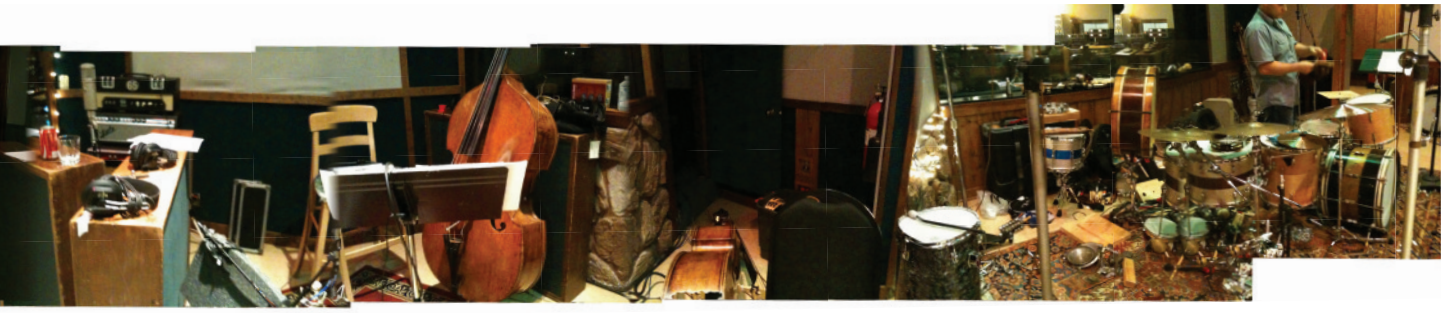
Despite Poole's obvious love for analogue, he also used Pro Tools for the *Band Of Joy* sessions. "I use Pro Tools all the time, because it has things that are incredibly usable. We can now do things that we could never do in the past. For the *Band Of Joy* project, we didn't only track to 16-track analogue, but we also went straight out of the Studer into a Pro Tools HD system, via an Apogee AD16. The Studer would have been set on input while recording, so the Pro Tools was receiving only the coloration of the Studer's transformers — the sound was not coming off tape — and I would be monitoring off Pro Tools.

"There were two different reasons for working like this. One was that we knew that we probably would need extra tracks for overdubbing and editing, and so we'd end up on Pro Tools anyway and would be mixing off Pro Tools. Monitoring through the system prevented any later surprises about the sound. We also didn't do any editing with razor blades — editing is so much easier in a DAW. I recorded to two-inch RMG 900 tape, at 30ips. We listened a couple of times to compare the tape and the digital versions, but we all liked the tape version better. So when we felt we had a good take and that we wanted to do an edit or just move onto the next song, I immediately transferred the take to Pro Tools, at 24/96. Once we had the analogue colour, it was just a matter of being pragmatic to work entirely in Tools

'Angel Dance' Input Signal Chains

Source	Mic	Preamp	Processing Chain
Mono drums	RCA speaker	Telefunken V76	Neve 33609
Bass drum	Neumann M147	Telefunken V76	Neve 1084
Overhead L	Coles 4038	Telefunken V76	Fairchild 670
Overhead R	Neumann U67	API 3124	
Room L	RCA KU2A	Neve 1073	Neve 2254, Pultec EQP1A
Room R	RCA KU2A	Neve 1073	Neve 2254, Pultec EQP1A
Bass amp	EV RE20	Neve 1055	Teletronix LA2A
Bass	DI	Neve 1084	Teletronix LA2A
D Scott amp	RCA MI6203	Neve 1055	UA 1176
D Scott mic	RCA MI6203	Neve 1055	UA 1176
B Miller amp	Royer R121	Chandler Germanium	
B Miller room	RCA 44BX	API 3124	
R Plant vocal	Avantone BV1	Telefunken V76	Inner Tube Atomic Squeezebox
Overdubs			
Tambourine	Neumann U47	API 3124	
D Scott BVs	Avantone CV12	Telefunken V76	UA 1176
B Bramlett BV	Telefunken AK47	Telefunken V76	UA 1176
R Plant BVs	Avantone CV12	Telefunken V76	UA 1176





from then on. If an overdub needed that analogue colour, I'd fly it over to the tape machine and then back into Tools. It's not as ideal as having it on tape to begin with, but it gets you part of the way there, sonically. Apart from a few backing vocal overdubs that I didn't fly back, almost everything on the album has touched analogue tape.

"The other reason for our recording setup was to be able to quickly do another take whenever needed. With Pro Tools I don't need to worry about how long the tape reel is, so I had it in record pretty much at all times. If we had reached the end of a reel, I could keep Pro Tools moving while rewinding or changing the tape, just in case the band was ready to record. One of the things that always happens in recording studios is that some magic is going on and the musicians look up at the control room with a look on their faces saying, 'Are you recording this?' I want to always be able to put my thumb up and indicate, 'We are rolling.' The musicians should never have to wait for the engineer. The sound of the format matters a little bit, but not as much as actually being able to be in record at the right time and capturing everything that's happening. I tried to roll the Studer for the actual takes, and Pro Tools was purely my failsafe that allowed me to keep things rolling at all times, just in case.

"With the Pro Tools system always recording, or set to record, the analogue equipment was my main focus during recording. I find that things are more intuitive for me when I'm working with analogue hardware. When I'm standing at a desk, my hands will reach intuitively for the EQ, and without even thinking about it I may at the same time do a fader move with my other hand. I can do all that instantly and purely on feel. By contrast, with the digital devices, even with big controllers, I find I need to think in a very linear fashion. It's very difficult to ride and tweak things at the same time, and you therefore end up putting small tweaks off. Sometimes you forget about them, or you may decide you don't have to do it anyway. So it really affects the way I work, and I think that of other people as well. Software forces you to interact with it on its own terms so much more than hardware does. It's like the old-style car radios: you could reach over and press big buttons and turn a big knob and dial in the station you wanted without having to look at the device. But with digital radios you need to look down and look for feedback from the device with every move you make. Your options are greater, but you have to pay much more attention, and this can distract you from what you should be doing, whether driving or listening to the music."

Non Standard

Homing in on the analogue equipment that was his focus at Woodland Sound, Poole elaborated on the recording stage of the album's opener and first single, 'Angel Dance'. Was his setup similar for all songs? Poole: "When Buddy and I work together it can be pretty free-form, and it's not uncommon for us to use different setups depending on what the instruments are, for example whether they're acoustic or not, and where the musicians need to be to give the best performance. By contrast, when I'm recording a project that I'm not going to mix, or for which the direction isn't quite clear yet, perhaps because the corporate side wants to have its input later on, I would normally have a fairly standard way of recording. With the *Band Of Joy* project, however, I felt free to vary the recording setup a little bit, as was appropriate for each song.

"We had a basic setup that we thought would be good for the bulk of the recording, which was that we had the drums and the bass in the main live room. Buddy was also in that room, but his guitar amp was isolated in another room, and Robert and Darrell had their own booths, in Darrell's case because he was mostly playing acoustic instruments. Everybody could see each other, everybody could come into the main room if they wanted; I remember we cut two songs with everyone except Robert in a circle, one of them being 'Central Two-O-Nine'. For me, it was matter of constantly adapting to what was happening, and sometimes I suddenly realised that people were going to do a take right where they had decided to sit down, which meant that I had to adapt my setup very quickly. In general, I set up quite a number of microphones, but did not use them all on each song. I also didn't have a standard drum-mic setup, because Marco had a non-standard kit and played non-standard parts. The room with Buddy's amp was large enough to have a room mic as well as a close mic."

Poole was organised enough for this interview to supply a copy of his assistant Gordon Hammond's input chain sheet, reproduced on the left. Rather than explain each detail, Poole elaborated on some of the less usual aspects of his setup, for instance the Avantone BV1 on Robert Plant's vocals, which went through a Telefunken V76 and then an Inner Tube Audio Atomic Squeezebox compressor. Poole: "We tried a few mics on Robert, but the Avantone was definitely the best-sounding. It gave us the picture that wanted of his vocals. I had only used that mic a little bit before, and it has definitely become a go-to mic for me. It is very good at tracking sibilance, although after the vocal had gone through a compressor or two by the time I mixed, I had to do quite a bit of detailed work on 'esses' and mouth noises and so on. The Atomic Squeezebox is also fairly non-standard, but it is a really good compressor. It reminds me a bit of the Summit Audio stereo compressor; it has the same feel but is a little bit more transparent. For the sound image we were after it was great. It also allowed me to put a highly dynamic performance on tape without the compression being audible. I always monitored through an [*Urei*] 1178, which was part of the mix path, and which gave the coloration we wanted to hear.

"The RCA speaker on the drums is a tiny speaker from before the days of computers, probably from a small cassette deck or reel-to-reel deck. I would find a spot in front »

» of the drums where there was a pretty good balance and I'd place it there. I used it to get a highly compressed, pretty nasty drum sound. I don't think I used a whole lot of it in the mix on 'Angel Dance'. The Neumann M147 did a great job on the bass drum, and the two RCA KU2A mics that I used as room mics are old ribbon soundstage mics known as 'the Skunk', because they have a large, six or seven-inch, black, ball-like windscreen with a white stripe down the back side, just like the stripe on a skunk. These were Gillian and David's mics, and they worked great on the room at Woodland, which does not have a lot of ambience, but the RCAs gave me a little bit of a different perspective than the overheads. The RCA MI6203 on Darrell's amp is a really mid-rangey mic, which worked great for this song. Buddy's amp had a Royer R121 going into a Chandler Germanium mic pre, which has a feedback knob that can be used as a bass tone control with the gain setting that I use. The Germanium is somewhat related to the Neve 1055; the circuits are similar, even though it doesn't quite handle the same.

"The Avantone CV12 on Darrell's and Robert's backing vocals is a sort of AKG C12-ish mic. It's a little bright for vocals, to my ears, but the BVs in this song were 'ahs' that needed some extra air, and I rolled the bottom end off anyway. I've also used that mic on upright bass and drums. You get a lot of bang for the buck with these Avantone mics! The RFT AK47 on Bekka's BVs is a new Telefunken and it sounds really good. It's warmer than the CV12 and probably a better overall mic. I also sent the input chain list of 'Silver Rider' [*see box, right*], and while many of the signal paths are the same, there are a few significant differences. 'Angel Dance' was recorded during the first two weeks of tracking, and 'Silver Rider' during the four day sessions in February. I changed the overhead mics to two Coles 4038s, which is a more classic setup, and the room mics to two Cascade C77 ribbon mics, which are a little more aggressive and worked better for this batch of songs. I also wanted a little more control on the floor tom on this song, which is why I threw on the Audio Technica ATM25."

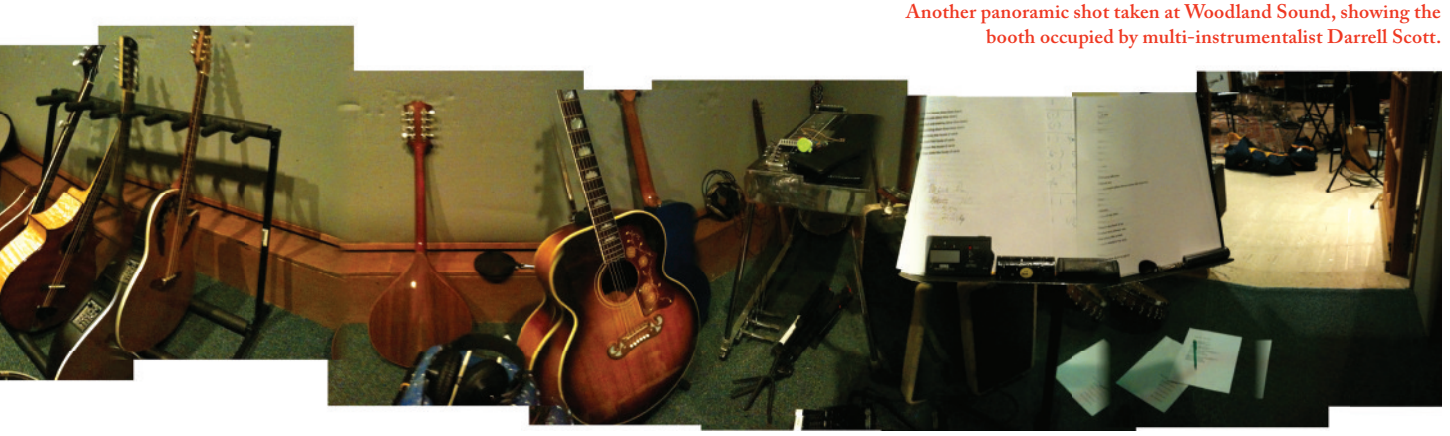
Out Of The Woods

"The console at Woodland Sound is lovely, but with only 16 channels and four buses, not big enough to mix. I need returns when I mix, not only for effects, but also for submixes, and compression and other effects. We went to House of Blues primarily for the sound of an analogue desk. I auditioned a few consoles for mixdown, including the big Neve at Ocean Way in Nashville, through that made the sound a little too warm and fuzzy. The recordings sounded great on the SSL at House of Blues: it had just the right amount of grit, along with clarity. Since so much is going on in the bottom end on this record, it was also very important to have monitors and a room capable of accurate low end. The Hidley-designed room and the Kinoshita monitors worked great for this. The SSL is very versatile in terms of EQ and dynamics, and the studio had all the coloration devices that I thought I might need: lots of API stuff, lots of Pultecs, lots of 1176s, lots

'Silver Rider' Input Signal Chains

Source	Mic	Preamp	Processing Chain
Mono drums	RCA speaker	Telefunken V76	Neve 33609
Bass drum	Neumann M147	Telefunken V76	Neve 1084
Overhead L	Coles 4038	Telefunken V76	Pultec EQP1A, Fairchild 670
Overhead R	Coles 4038	Telefunken V76	Pultec EQP1A, Fairchild 670
Room L	Cascade C77	Neve 1073	Neve 2254, Pultec EQP1A
Room R	Cascade C77	Neve 1073	Neve 2254, Pultec EQP1A
Mono drums	Sony C37	Telefunken V76	Neve 33609
Floor tom	Audio Technica ATM25	API 3124	
Bass amp	EV RE20	Neve 1055	Teletronix LA2A
Bass	(DI)	Neve 1084	Teletronix LA2A
D Scott mic 1	RCA MI6203	Neve 1055	UA 1176
D Scott mic 2	RCA MI6203	Neve 1055	UA 1176
D Scott amp	Shure SM7	Neve 1073	
B Miller amp	Royer R121	Chandler Germanium	
B Miller room	RCA 44BX	API 3124	
R Plant vocal	Avantone BV1	Telefunken V76	Inner Tube Atomic Squeezebox
P Griffin vocal	Telefunken AK47	Telefunken V76	UA 1176
Overdubs			
BVs L	Neumann U67	Telefunken V76	Urei 1178
BVs R	Neumann U67	Telefunken V76	Urei 1178

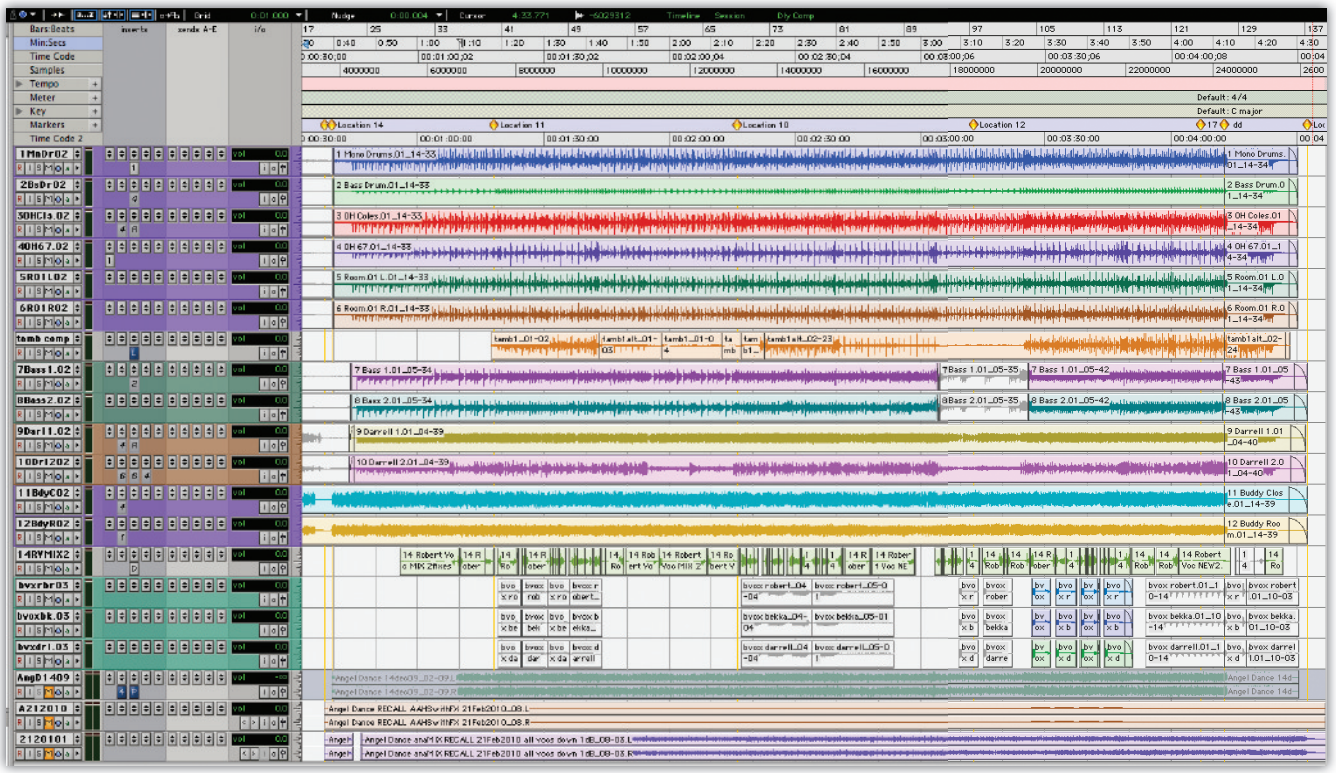
Another panoramic shot taken at Woodland Sound, showing the booth occupied by multi-instrumentalist Darrell Scott.



of Neve modules. Another factor was that since I knew that we still had to do some overdubs, having a large board enabled me to set a mix up on one part of the board, and use another set of fader banks for overdubbing on another song. I rarely had more than 24 tracks of music to mix, plus another 20 returns. This meant that I had almost 40 extra faders to use for the overdub mixes.

"Regarding my mix method, it really is mainly about getting a direction for what's going on. You push up the faders and figure out what the dominant vibe of the record has to be, like: this is a song where the electric guitar really needs to dominate, or this song is about the bass line, or in this song the voice needs to be inordinately loud. Once that's defined, it's a matter of figuring out what the other instruments need. Do the drums take up too much space, do they need to be more compact and darker? Does that acoustic guitar need to be bigger, or smaller, or more mid-rangey? I work out the colour and general sound of each instrument, and might patch in three different compressors, or three different EQs, to see which one places the instrument in the right spot.

"Because I had recorded the music, and because we had been working with a 'We do it the way we like it and we make our decisions early' mentality, I didn't have to do that much in the mix. This is in contrast to when I'm mixing a track recorded by someone else, or for which the direction was to be decided during the mix. With regards to *Band Of Joy*, the whole idea of the album was to have an acoustic feel, yet make it gritty. We didn't strive for hi-fi or a clean sound or things like that. We all like stuff that's a little more in the gut, rather than just tickling your ears. So there »



The entire Pro Tools Session for 'Angel Dance'. None of the plug-ins were used in the final mix, and the edits on Robert Plant's vocal track (coloured green, about two-thirds of the way down) were mainly to tackle sibilance issues. At the bottom you can see mixes were recorded back into Pro Tools from the SSL desk.

» were things that I did to make it sound more gritty, especially on Darrell's acoustic instruments, like running his mandolin track through a mic pre to get some distortion on it.

"You can see the effects I used on my mix notes [*right*]. The effects at the bottom of the sheet — Roland RE501, Lexicon PCM42, Yamaha SPX90, TC Electronic D•Two, Roland SRV2000, AMS DMX15, AMS RMX16 — are going into my sends and returns. The RE501s, PCM42s and D•Two would all have been set to a delay, not set to a precise tempo, but definitely something that accentuated the groove. The House of Blues did not have a real plate for a long reverb, so I used the cheap SRV2000 for that: if you massage it the right way, you can get something that works really well. The DMX15 delay works as a sort of smearing device for things you don't want to be able to aurally locate as a point source. I tend to use the RMX16 for a short ambient reverb.

"The SPX90 did a pitch-chorus thing, its main advantages being that it doesn't sound too hi-fi, and that it's fast to set: five parameters, and two of those you don't normally have to touch. I'd rather spend 10 seconds adjusting an SPX90 than having to pull up a display and go through four pages of menus on a higher-end device. I didn't use any plug-ins on this session — the ones marked in the screenshot were used for rough monitor mixes during recording, but I disabled them during the final mix. There are some plug-ins that I really like, but I have yet to hear a plug-in that emulates a specific analogue device and sounds as good. But if you're mixing in the box, UA make some really good-sounding plug-ins, and the GML EQ plugs are one of the few plug-ins I've used that behave like their hardware equivalents. Of course there are some plug-ins for which there are no analogue equivalents. And in the end, the effect that you have, whether outboard or analogue, is always better than the one you don't have!"

Drums: Shep 1073 EQ, Chandler EMI TG12413, SSL desk compressor, Pultec EQ, Roland RE501.

"I used the Shep 1073 EQ on the overheads, which is a box that looks exactly like the Neve 1073, except it's made by Shep. It sounds really close to the original. The only compression I had on the drums was a Chandler reissue of the EMI TG12413, which was on a bus. All the drum tracks went through that, and I also occasionally used the SSL compressor on individual channels. I also had Pultec EQ on the drum bus, I prefer their coloration to the SSL EQ on drums. I don't think there were any delays on the drums, apart from maybe a bit of 501, while the ambience came from the room tracks."

'Angel Dance' Mix Notes

Source	Tape Out> Line In	Insert
Mono drums	API 560	
Bass drum		
Overhead L	Shep 1073	
Overhead R	Shep 1073	
Room L	API 560	
Room R	API 560	
Tambourine		
Drum extra bus	EMI T61, Pultec MEQ5	
Drum extra bus	EMI T61, Pultec MEQ5	
Bass subgroup		UA 1176, Pultec EQP1A3
Bass amp		
Bass DI	API 550	
D Scott amp	Pultec MEQ5	
D Scott mic	Empirical Labs EL8, Neve 31102	Neve 1081
B Miller amp	Neve 1081	
B Miller room	Neve 1081	
R Plant BVs		
B Bramlett BVs		
D Scott BVs		
Vocal extra		
R Plant vocal	Urei 1178, API 550	
Vocal FX send	LT Sound CLX	

Auxiliary effects: Roland RE501, Lexicon PCM42, Yamaha SPX90, TC Electronic D•Two, Roland SRV2000, AMS DMX15-80S, AMS RMX16.

Bass & guitars: Universal Audio 1176, Pultec EQP1A3, Empirical Labs EL8 Distressor, Neve 31102.

"I grouped the bass amp and bass DI track together and put a UA 1176 and a Pultec EQP1A3 on them. The Empirical Labs Distressor EL8 going into a Neve 31102 mic pre on Darrell's mic track was an example of me adding distortion because the track sounded too clean. I used the Distressor to lower and then raise the signal up to line level and get the right amount of crunch going. Darrell's stuff was acoustic, so any reverb or delays that you hear on them came from me. Most of the effects that you hear on Buddy's guitar come from him. I might have added a bit of delay in the opposing speaker, just to bring a guitar across the sound field a bit more."

Vocals: Urei 1178, API 550, LT Sound CLX, Roland SRV2000 & RE501, AMS RMX16, TC Electronic D•Two.

"I had the Urei 1178 compressor and the API 550 EQ on Robert's vocal, along with a parallel channel that was even more compressed. So any time he hit a low note that sounded a little muddy and needed some more clarity, I pushed up the parallel channel. There's also an effects send channel with the LT Sound CLX on it, which is a Dbx VCA compressor with a good de-esser in it, and that was used as the send to any effects devices. As a side note, all the little slices you see on the screenshot were not performance edits, but mostly me trying to get the sibilance right. I often take out little clicks and pops and also 'esses' or 'effs' out manually. A de-esser will hit everything in a technical-sounding way, and sometimes you may want to retain the emotive way a certain 'ess' or 'eff' is expressed, but only precisely change the gain, just to correct recording artifacts. 'Angel Dance' is one of the few tracks on which Robert's scratch

[*tracking*] vocal didn't make up the largest component of the final vocal track. On this song, he re-sang the vocals later, and I think he did two passes and we used mostly one, with a few lines from the other pass thrown in. For reverb on his vocals, I used the SRV2000 long reverb and a little bit of RMX16; for delay, either the D•Two or an RE501. I used very little reverb on the tracks on the album in general, apart from on Robert's vocals and the backing vocals. The other ambience you hear may have been a long delay from the 501 or PCM42 or any of the room mics."

Mike Poole at House of Blues Studio, where *Band Of Joy* was mixed. Producer and guitarist Buddy Miller is in the background.

Mixdown

"You can see on the screen shot that there are multiple mix channels at the bottom. I'll normally mix directly back into Pro Tools, either through an Apogee PSX100 or the Burl Audio B2 Bomber A-D converter. We got the latter halfway through the project and we actually felt that it's superior to the PSX. We also went to tape, to an Ampex ATR102, half-inch tape at 15ips; 15ips has such a different tonality than 30ips, and because this record was so much about the bottom end, the extended top end of 30ips wasn't helping us at all, while we liked what 15ips did to the low end. We compared the digital and tape mixes and ended up using the tape version. I had a bit of SSL bus compression over the stereo mix, and that was it. I think it came out pretty well!"

With that conclusion, Mr Poole, very few will have an argument...

Prominy SR5 Rock Bass

Virtual Bass Guitar Instrument

Virtual bassists are not short of options for looking after their low end. How does this new one fare?

NICK MAGNUS

Japanese company Prominy are well known for their SC and LPC Electric Guitar virtual instruments. Their latest product, SR5 Rock Bass, drops down a couple of registers to join a growing legionette of virtual bass-guitar instruments hosted by the ubiquitous Kontakt Player (see the 'Alternatives' box). At the heart of SR5 is a 24-bit, 15GB core sample library of approximately 14,000 samples, expanded from 8GB of real world data using NI's .NCW lossless compression format. As with Prominy's virtual guitar products, the aim is to recreate authentic-sounding bass-guitar performances, both in terms of sound and of detailed performance techniques, as easily as possible, through live performance on a MIDI keyboard. The bass guitar Prominy have chosen to sample for SR5 is a Musicman Stingray, notable for being a five-stringed instrument with a range going all the way down to a low 'B'.

The full SR5 bass guitar, showing the new Global GUI tool, fretboard monitor, the performance view for the 'string 1' main instrument, the remaining main instruments (minimised) and the first few 'add-on' instruments.

How It Works

Following a similar format and concept to the SC and LPC Electric Guitars, the complete, fully detailed SR5 Rock Bass instrument is constructed from 14 separate patches within a Kontakt Multi, all running on one MIDI channel. The patches fall into two categories; 'Main' instruments (the five strings, each having its own patch) and keyswitchable 'Add-On' instruments (harmonics, release noises, scrapes, slides and other distinctive bass-guitar noises). There are four playing articulations available to the Main instruments: legato slide (sliding between notes when played in a legato manner, with three selectable slide speeds), single (no slide), hammer-on/off, and repetition (notes retrigger when released). These are supplemented by muted and picking articulations, controllable optionally by mod-wheel position or velocity range. While certain events are handled 'intelligently' by KSP scripting (for example, string selection) the majority of sounds are activated manually by keyswitching. However, string selection can also be overridden using keyswitches to force notes to play on specific strings if required. SR5 plays in a monophonic, legato mode by default; once a note is played, its sound continues either until another note is played, or until the sample reaches its natural conclusion. Alternatively, notes can be terminated using 'stop' and 'hold' keys. These each trigger different release noises: fret slide, downward glissando or pick noise, for example. The user can customise these release noises if they choose, re-assigning their locations or even disabling them altogether.

Unlike many virtual guitar instruments, SR5 has no 'strumming' feature, hence no strumming trigger keys. However, the instrument can be played polyphonically by disabling the default monophonic 'auto sustain' mode, either via a MIDI CC command or by pressing the MIDI sustain pedal. The full instrument also includes »

Prominy SR5 Rock Bass €95

PROS

- High-quality, detailed and lively sound.
- Well suited to rock, metal and progressive genres, also adaptable to many other styles.
- Keyswitch functions are easy to learn.

CONS

- Demanding on RAM, especially with Kontakt versions below 4.1.

SUMMARY

A lively-sounding and very playable virtual bass guitar instrument featuring many controllable, characteristic bass-guitar artifacts. Although its tone is optimised for all forms of energetic rock, given suitable treatment it also adapts well to many other genres.

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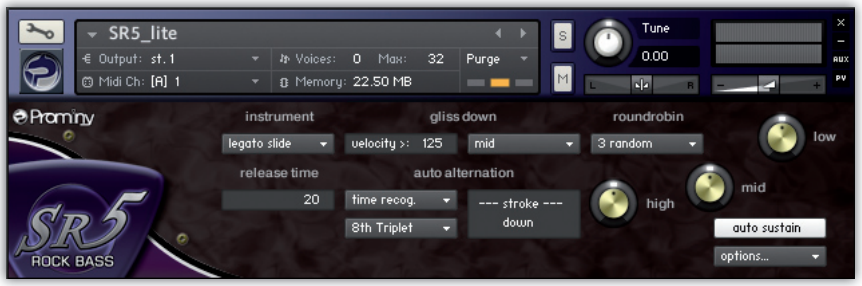
The SR5 Library

SR5's library is organised very simply. The Multis folder contains only two patches: the complete, fully-articulated instrument, SR5_Rock_Bass.nkm, described above, and a 'lite' version, unsurprisingly named SR5_Lite.nkm. For situations requiring less detailed control, the 'lite' Multi presents SR5 in a slightly less RAM-hungry cut-down form, as a single patch. In this version, the key mapping of the five strings is fixed, with no option to 'force'-select specific strings (thereby requiring considerably fewer samples). The main articulations (legato slide, single, hammer-on, repetition, mute and pick) are all available as before, plus six release noises assignable to the Hold and Stop keys. This lite version is nevertheless very expressive and capable of very convincing results.

The Instruments folder contains the individual Main and Add-on instrument patches. These can be loaded as required in any combination to construct a custom Multi suitable for the application you have in mind.



The cut-down SR5 Lite version is contained within one Kontakt patch, yet still provides the main playing articulations and enough detail to create a convincing bass guitar performance.

Look Lively

Prominy have sampled their Musicman bass with great clarity; the samples have oodles of sustain, and plenty of top end to work with. Plectrum strokes are clearly defined, and there's just the right amount of buzzes, fret rattles and other details to give interest and a sense of urgency to the sound. In order to avoid the dreaded machine-gun repetitions (a vital consideration in rock and metal styles where rapid note picking is typically commonplace), round-robin sample randomisation brings a degree of 'natural' tonal variety, with up to four cycling modes available. Settings '3' and '4' deliver increasingly wilder tonal variations, with the result that heavy emphasis is frequently applied at inappropriate moments. This is most noticeable when SR5 is heard 'clean', although the inconsistencies become less obvious with compressed, overdriven amp settings. If you're after a more even tone, the '2' setting is the most effective. Round robin can also be disabled if you need a super-consistent 'workstation' bass sound. Even then, the sound isn't entirely lifeless, as SR5 features up/down plectrum strokes with automatic stroke detection This syncs to your song's tempo at a variety of selectable note values, adding further to the sense of animation and tonal variation. Plectrum strokes also can be forced to play in one direction for a more even, emphatic sound.

Conclusion

Despite the seemingly complex keyswitch learning curve, SR5 is, in fact, very easy and quick to learn. Its liveliness of tone makes it ideal for all forms of energetic rock music when treated through a good bass-amp simulation, and I even found it capable of producing the classic 'belching' sound beloved of prog rock bands — a sound I've tried many times to emulate in the past, but never as closely as this. Despite its patently rock and metal intentions, I found SR5 worked perfectly well in a variety of genres — although you may have to massage your amp settings accordingly to make it fit into more gentle styles of music! ■■■

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Play While Loading: The RAM-ifications

Heads up, Kontakt 4 users — if you don't already know this, you're going to want to. The following may seem off-topic with regards to SR5, but in fact it's extremely relevant. Not only does the update to version 4.1 load instruments faster than before, it also brings an excellent new facility, Play Whilst Loading. This feature enables even large, 'Giga'-sized instruments to be played immediately from the moment the instrument appears in the Kontakt GUI — even huge instruments that would normally take several minutes to load. If you play a note whose sample has not yet been loaded, Kontakt streams it from disk on the fly, whilst continuing to load the rest of the instrument.

This is very convenient for those moments of great haste or impatience, but it also provides one truly welcome benefit for those of us struggling to run multiple monster-sized 'RAM Hog' Kontakt instruments on a 32-bit PC with the maximum 3GB of accessible RAM. The problems typically arise when your RAM is already chock-a-block with sampled instruments, but there's still that killer Dagenham Girl Pipers Kontakt instrument that you simply have to include as the icing on the cake. Trouble is, you only need half a dozen notes from it, but to access those, you're obliged to load the entire instrument. Frustratingly, our Girl Pipers are far from being size zero — they need to gobble up 600MB of RAM! Normally, a workaround might be to render everything else to audio to free up memory, but what a pain that is when you want to keep everything running 'live'. Not any more — simply follow this indispensable tip from the Dr Magnus clinic. In a spare 10 minutes when you and your computer are otherwise idle, load one of those RAM-hungry instruments into Kontakt. When loading is complete, open the instrument's Purge menu, and select Unload All Samples. You'll notice there is now a red indicator showing no samples in RAM. Ordinarily, you'd expect to hear nothing when you play any notes, but now, when you play your MIDI keyboard, there they are, streamed instantly from disk, loading on demand. Now all you need to do is make sure all samples are unloaded once more, and re-save the 'empty' patch under a new name. Next time it's loaded (which happens very quickly) no RAM will have been consumed at all. Only the notes you actually play from that point onwards are loaded into RAM, and you'll find your Dagenham Girl Pipers part will probably only consume about 10MB, leaving loads of room for other instruments. So as far as SR5 is concerned, you should always be able to take advantage of the full instrument with all its articulations, if you follow this simple preparatory procedure.

Alternatives

Native Instruments produce a highly respectable and growing range of bass guitar instruments that includes the **Scarbee MM-Bass**, **Pre-Bass** and **Jay-Bass**. Other front-runners include **Trilian**, **Chris Hein Bass**, **Liquid Electric Bass** and **Vir2 Basis**. Of these, only Trilian uses a proprietary sampler engine as opposed to being hosted by Kontakt and the ubiquitous Kontakt Player.

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Tonehammer Requiem

Choral Sample Library

Sanctify your soundtracks with Tonehammer's new choral library.

DAVE STEWART

Over the last year or so, West Coast company Tonehammer have been steadily making their mark on the sample library market with an idiosyncratic and creative combination of exotic percussion, thunderous drums, moody pianos, evocative female solo vocal collections and er, sampled bees. Their latest release is by far their most ambitious, featuring the combined talents of 37 singers and a world-famous choir conductor. Recorded over a period of two weeks in the hallowed acoustic of one of San Francisco's oldest cathedrals, the choir was conducted by Robert Geary, artistic director of the acclaimed Volti choir (who specialise in contemporary music by US composers) and the 200-voice San Francisco Choral Society. SFCS's Alan Kleinschmidt was also on hand to direct proceedings.

Portentously titled Requiem, the library includes a full choir of sopranos, altos, tenors and basses, separate male and female groups, and five soloists (two sopranos, tenor, baritone and bass), all recorded 24-bit from multiple microphone positions. It ships in two versions: Requiem Pro (which requires the full version of Kontakt 4) is the master library featuring all the mic positions. Optimised for use on laptops and smaller systems, Requiem Light has a reduced sample and articulation menu, includes free Kontakt Player software and blends the different mic position samples into a single stereo mix. Upgrades in both directions are available — the Light brigade can turn Pro by shelling out more dosh, and Pro users can purchase the Light stereo samples as an add-on.

Tonehammer have a download-only policy. Though I applaud this conservation-minded approach, there are potential pitfalls: you'll need a fast broadband connection to download Requiem Pro's 23GB of samples, and once they're copied onto your drive you obviously can't send them back. *Caveat emptor* applies here, so to avoid any misconceptions over what you're buying, be sure to listen to the demos of Requiem at Tonehammer's site beforehand.

Anyone minded to make pirate copies of the library might want to think again, as the makers have individually watermarked each sample and can thus trace unauthorised copies back to the original owner. (Cue police car siren.) However, Tonehammer are happy for you to freely duplicate the library on your own system without a dongle — they just don't want you to give away the fruits of their labour to others. It's hard to argue with that.

The Gothic hymn-book graphics of Requiem's user interface conceal a secret...

Cathedral Tones

The choir was recorded from three main mic positions, stage, mid and far, with microphones placed respectively five, 15 and 45 feet from the singers. The soloists were also close-miked, giving the option of a drier, mono-ish sound. Certain performances segregate the male and female singers, but rather than

Tonehammer
Requiem Pro **$799**

PROS

- A top-notch classical choir performing a wide choice of articulations and effects in a fab cathedral acoustic.
- Interval-based legato samples produce excellent joined-up melody lines.
- Contains handy phrase-building tools.
- The multiple mic positions are great for surround mixes.

CONS

- Dynamics are limited to piano and forte.
- Why do classical singers always roll their 'r's in that annoying way?

SUMMARY

Whether you're scoring a Satanic chiller, a Vatican conspiracy thriller, an epic fantasy top-of-the-biller or Alex Reid's walk-on music, every film, TV and game composer needs a good sampled choir in their locker, and this one fits the bill perfectly.

Requiem Articulations

- Multi-vowel sustains (m, w).
- Legato sustains (fc).
- Classical Latin words (fc, m, w, s).
- Marcato syllables (fc).
- Staccato syllables (fc, w).
- English words (fc).
- Numbers (m).
- Consonants (fc).
- Slow swell woo — aah (fc).
- Slow trill (fc, m, w).
- Dissonant sweeps (up/down/fast) (fc, m, w).
- Clusters (long/shrill) (m).
- Improvised 'tone ambiences' (fc).
- Body noises (fc).

(fc = full choir; m = men only; w = women only; s = soloists.)

A detailed list of Requiem Light's reduced menu of articulations can be read at www.tonehammer.com/docs/tonehammer_requiem_light_readme.pdf.

sampling sopranos, altos, tenors and basses separately, the choir is mainly presented as a full unit, with the different voices mapped and blended according to range. This has been done so skillfully and naturally that I never noticed the transitions, even when moving between the high male register and the female altos' range.

As well as providing the basic 'aah', 'ooh' and 'eeh' sustained vowel sounds traditionally found in choir libraries, Requiem adds around 10 subtle variations extracted from longer performances sung by separate male and female groups, and also throws in 'eh', 'ih', and 'um' options (the latter sung by the men only). The sustained multi-vowel presets are recorded at tone intervals, looped, one-dynamic and perfectly in tune; the men's deliveries have a triumphant ring while the women make a more intimate sound, but both retain a full-bodied, warm and engaging timbre throughout. I noticed that the precise, wide-stereo detail of the stage miking becomes less evident in the 'mid' position and disappears altogether in the far miking, which nevertheless has a nice stereo image and a lovely ambient halo. Pan these 'far' samples to the rear speakers of your surround mix while positioning the stage miking at the front and your listeners will be effectively seated in a San Francisco cathedral — better warn them not to swear.

Each multi-vowel preset has two layers, which can be crossfaded with an expression (CC11) pedal or controller. Crossfading quickly back and forth between 'ah' and ooh' creates a diphthong effect that the brain translates as 'oowah-wah-wah'. The San Franciscan choir members might not be best pleased to learn that their vocal efforts can be transformed into a virtual wah-wah pedal in this way, but your reviewer found it to be a lot of fun. Experiments with other vowel combinations threw up some interesting possibilities; in tandem with the ability to instantly keyswitch both layers' vowels, this adds a realistic and expressive mobility to what would otherwise be tonally static samples.

Tonehammer gently warn that while a historic stone building with acres of stained glass and huge wooden pews creates a stunning visual and sonic environment, the temperature changes caused by the hot Californian midday sun occasionally caused the pews to creak. I heard nothing »

Click on the hymn book and an elaborate performance control panel (embossed on alchemist's parchment, naturally) magically opens. What sorcery is this?

» troublesome while auditioning the samples, but if you detect the odd extraneous noise you can console yourself that it's a natural acoustic phenomenon occurring in a sacred place, rather than a digital spike caused by some malfunctioning piece of kit.

Alphabet Soup

Those interested in choral 'word building' will be pleased to hear that Requiem gives you the means to create virtual phrases. The easiest way to do it is simply to play the classical Latin words; these were sung by the choir at a choice of fast and slow speeds, and presets with 'time-stretch' in their name use the pitch wheel to control speed. The ancient language took me back to my school days — as I heard the familiar chants of 'Adoramus', 'Crucifixus', 'Dominum', 'Lumine' and the rest, I could feel myself slipping into the near-coma brought on by interminable Latin lessons on wet Thursday afternoons. O Taedium, Miserere Mei... Fortunately, the phrases are sung with sufficient intensity (particularly by tenor Michael Mendelsohn, who hisses his esses and rolls his 'r's with great pantomime-villain gusto) to keep us awake, and mercifully I'm no longer required to translate them. In an understandable outbreak of self-big-upping, the makers also asked the choir to improvise renditions of the non-Latin word 'Tonehammer', which produced some fairly alarming results.

The library lets you create your own words via 19 different Latin-esque consonant-vowel combinations, sung in staccato and longer marcato styles: Ag, Cre, Cru, Do, Fah, Fis, Glo, Ky, La, Mus, Nis, Nus, Rhi, Rru, Sanc, Sart, Sin, Son and Tus. I eagerly set about arranging these into insulting phrases aimed at our hapless coalition government, but managed only 'Fahtus Sartus'; although that had a satisfyingly scatological ring, it failed to accurately convey the full extent of my fury. These full-choir syllables are sung at maximum volume (I should think so too — if you're going to sing 'ag', you might as well bellow it at the top of your voice). Regardless of their syllabic content, it's good to have such blasting, energetic performances in your arsenal.

Alternatives

EastWest/Quantum Leap's Symphonic Choirs was the first choral library to feature multiple mic positions and a 'word building' utility, while the more recent **Vienna Choir** maintains **Vienna Symphonic Library's** reputation for forensic musical detail and superb legatos. Both feature separate sopranos, altos, tenors and basses, and **EWQLSC** also offers soloists and boy's voices. First issued when 32MB hardware samplers ruled the earth, **Spectrasonics**' five-CD **Akai** library **Symphony Of Voices** has a similar line-up to **EWQLSC** and includes pop-style 'oohs' and 'aahs'. Back in the present day, the impressive specs of **Cinesamples Voxos: Epic Virtual Choir** make it a worthy contender.

To help join up these disparate sounds, Tonehammer supply a couple of easy-to-use, fun tools: the Quick Chant Builder functions like a classic TR808-style step sequencer: set the sequence length in bars, specify a tempo in Kontakt and (if necessary) a tempo multiplier, then click on a grid to select the timing and velocity of whichever syllables you wish to trigger. The Phrase Builder does a similar job, and its hilarious graphic of a flame moving across a row of unlit candles is worth the entry price alone. Rather than playing back a sequence of syllables when you hold down a key, the Phrase Builder advances to the next one each time you play a new note. A maximum of 16 syllables can be used, and you can insert blanks if you wish. I tried to perform a version of the spine-chilling legend emblazoned on the walls of Gaudi's Sagrada Familia: 'Sanctus, Sanctus, Sanctus' — it didn't sound quite right because the 'c' of 'Sanc' is enunciated too softly, but 'Agnus Agnus Agnus' ('sheep

System Requirements & Installation

Requiem is formatted for Native Instruments' Kontakt 4 sampler, which runs stand-alone and as a plug-in on Macs and Windows PCs. The Pro version runs on the full version of Kontakt 4.1.1 or later, and will not work with Kontakt 3.5 or lower, and the Kontakt Player supplied with Requiem Light will not play the Pro instruments. Requiem Pro requires 22.3GB of disk space, while the Light version needs only 3.2GB. Tonehammer don't publish minimum specs, but as with any library utilising large numbers of samples and multiple mic positions, a fast computer with plenty of RAM is advisable — I'd recommend a minimum of 4GB. If your system has less, you can still use the library effectively by sticking to one mic position, using the optional DFD versions of instruments and utilising Kontakt's 'purge' function to remove unused samples from your arrangements.

Buyers are sent an email containing a download link to a set of compressed RAR files (the Pro version has over 30 of them — if you have a slow broadband connection, they'll take a very long time to download). There are separate RAR files for the samples, the Kontakt instruments, the user interface images and a small utility for installing the last. After downloading, you extract the files' contents with UNRAR, a utility which most people have in their systems. If you don't have it, Tonehammer tell you where you can download it free of charge. You only need to extract the first sample's RAR file (the rest will follow automatically), and the makers supply clear, detailed instructions about where to place the extracted data on your hard drive. After that, it remains only to authorise the library at the Native Instruments Service Centre, using your serial number.

sheep sheep') sounded spot-on.

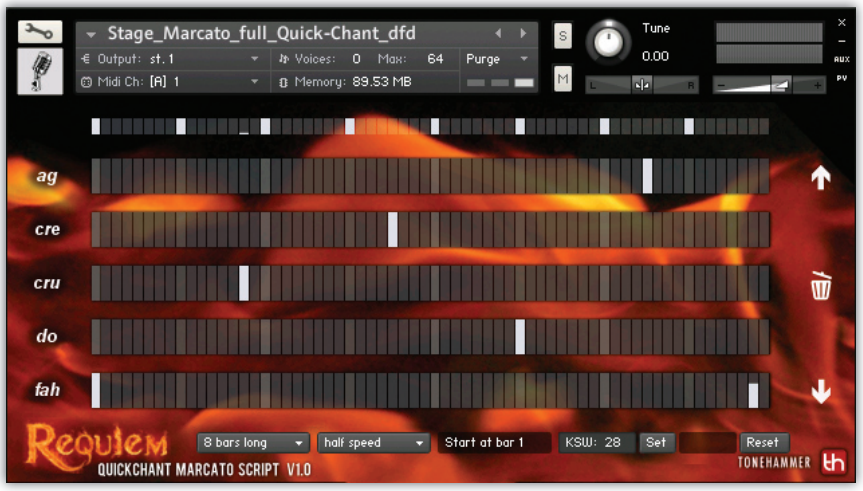
If you're feeling adventurous, you can create quasi-syllables by linking a vowel sound to one of the 54 simple and compound consonant sounds included in Requiem. I used to be sceptical about constructing 'words' with samples, but must admit that things have now moved on to the point where it sounds pretty realistic. Tonehammer's efforts in the field are to be applauded, and I'm sure composers will appreciate being able to add lyrics to their melody lines, even if they are complete nonsense!

Smooth Ride

To achieve the all-important 'joined-up' effect of real-life legato performance, the producers implemented script-based interval detection, which calculates played intervals in real time and inserts a tiny transitional 'interval sample' between notes. The legato samples are long sustains based on the vowels oh, ah, and eh, sung at two dynamics; they default to monophonic, but can instantly be rendered polyphonic (and non-legato) for chordal passages by the use of the 'legato off' button.

Requiem's Phrase Builder syllable-sequencer, complete with moving candle flame graphics. (Monks not provided.)

The transition samples' response is governed by an on-screen speed setting: the 'slow' setting produces gentle, smooth transitions, while 'slow', 'mid', 'fast' and 'tight' incrementally shorten the transitions, making the samples more suitable for faster playing. I found the system worked very well for slow and mid-tempo passages, but couldn't cope with very fast lines or trills. (Tonehammer admit as much in their documentation.) A trill sung by 37 vocalists would sound pretty weird anyway, so I was happy to accept the limitation and wallow in the sound of the voices performing beautifully smooth, stately legato lines.



The 'Quick Chant Builder' in Requiem works like a classic step sequencer, allowing you to control the tempo, rhythmic placement and volume of syllables in word-building sequences.

Supplementing the multisamples, words and syllables is a large menu of excellent vocal effects. Dissonance is a strong point, with the mens' exaggeratedly nasal cluster chord delivery taking first prize in the 'Sounds You Will Never Hear On *The X Factor*' competition. Claps, finger snaps, foot stamps, shouts, whispers and breath effects are included, along with some rarely-found 'teeth chattering' performances that samplists have been seeking for years. The effects range from funny to frightening, and are of sufficient quantity and quality to satisfy the most ardent experimentalist.

For those who enjoy delving deep under the bonnet, Tonehammer's tech team has provided a host of facilities too numerous to list here. Not everyone will have the time and commitment to explore them, but those who do will find their patience rewarded by some helpful and subtle performance-shaping tools. To get an idea of the library's deep programming capabilities, I recommend you download the Requiem Pro version manual from www.tonehammer.com/?page_id=4276.

Mindful of the heavy RAM usage of some of their presets (the three-way 'ah/oh/eh' legato instruments are particularly demanding), Tonehammer created DFD (direct from disk) versions of them all to reduce the strain. These presets rely heavily on disk streaming and therefore may require higher soundcard audio latency settings to avoid drop-outs. This is a common trade-off with such libraries, and may be the best way to go if you need to build complex choral arrangements on a smaller system.

Conclusion

We may live in irreligious times, but there's still something transfixing about hearing massed voices intoning ancient Latin phrases. Requiem's samples are highly effective at producing an instant aura of sanctity, and also pack enough power to cut through a full orchestral arrangement. Adaptable, subtle, expressive, often beautiful and occasionally forceful, these are choir samples of the first order, and the multisamples and effects are a highly desirable addition to a MIDI composer's resources. If the price of the Pro version seems steep, you can save yourself a considerable amount of money and still retain the essence of the library by buying the Light version; either way, when you consider the amount of work and expertise it takes to create a collection of this stature, Tonehammer's pricing seems very reasonable. ■■■

£ Requiem Light $299, Requiem Pro $799, Light to Pro upgrade $599, Light add-on for Pro owners $99.
E info@tonehammer.com
W www.tonehammer.com

PLUG-IN FOLDER

Toontrack EZ Mix

Formats: Mac & PC VST & RTAS, Mac AU

Toontrack seem to be on a mission to make everything EZier. First they made it possible to record more or less convincing drum tracks without having to understand the mechanics of drumming. Now they are attempting to make it possible to mix without understanding the mechanics of mixing.

EZ Mix has been developed by Overloud, and is, at least in theory, the ultimate 'one size fits all' plug-in. Its user interface consists of three faders labelled Shape, Blend and Level, plus a large preset browser. The 200-plus presets are categorised by name, instrument type and effect type — or by the name of the programmers who created them, should you belong to a parallel universe in which that might be useful — and they can be sorted alphabetically in any of these categories, marked as favourites, and found using free-text searches.

The idea is that you insert EZ Mix across an instrument track, then browse the presets that suggest themselves as being useful for that instrument. Once a preset is loaded, the Shape and Blend sliders adjust appropriate parameters within that preset. So, for example, on the 'Chorus Guitar' preset, the Shape slider controls chorus speed, while Blend adjusts the wet/dry mix of chorus and reverb simultaneously. Others are more complex: on '12BitReverbwithFilterDelay1', for example, the Shape slider controls reverb time, delay time and feedback all together. The presets cover the full gamut of effects and processes, from simple compression and EQ to more unusual processes such as bit crushing, filtered delays and the proprietary 'Overloud' — frequently in combination.

In practice, EZ Mix is sometimes more versatile than you'd expect. For instance, I was wondering how on earth any of the gated drum presets could possibly work, given that there is no user control over gate threshold, but in practice they seem to use an intelligent algorithm that can pick up wanted audio and exclude spill fairly effectively. I inserted an instance of EZ Mix over every track in a multitrack drum recording, chose appropriate presets, and was surprised at how good the results were. It sounded like a mix, and it was easy. Job done. Provided the original tracks were fairly well recorded, it likewise provided plausible bass, guitar and vocal sounds, though, naturally, they were somewhat generic.

I was less taken with EZ Mix in some other roles, though. For instance, there are 20 or so reverb presets designed for use on an aux send; I didn't really warm to any of them, and two faders really don't provide enough control for a reverb.

For those with no time or curiosity, EZ Mix does what it says on the tin. It's also very affordable, and expandable through the addition of preset packs that will be sold separately. But if it's aimed at newcomers to mixing and recording, I think it's rather a shame that it isn't possible to lift the lid of the tin a little. The ability to inspect — and adjust — presets in detail to see what goes into them could be an excellent learning aid. As it is, it certainly offers an instant route to a more polished sound, but I suspect many users will outgrow it quite quickly.

Sam Inglis

£ **£45 including VAT.**
W **www.timespace.com**

> **"I inserted an instance over every track in a multitrack drum recording, chose appropriate presets, and was surprised at how good the results were."**

2C Audio Aether 1.5

Formats: Mac & PC VST, Mac AU

Back in *SOS* September 2009, I reviewed 2C Audio's first product, the rather splendid Aether reverb, whose algorithmic excellence left me deeply impressed. You might expect that this upgrade to version 1.5 would simply offer some bug fixes and the odd tweak, but in fact this is a significant upgrade incorporating many new features, including native 64-bit versions (RTAS and Pro Tools are also promised in the near future).

The original Aether interface proved to be a 'love it or hate it' design, so I'm happy to report that the new and more conservative default GUI of version 1.5 offers significant improvements to legibility, with clearer control groupings, easier to read parameter values, and photographic renditions of all the early-reflection space options rather than the somewhat cryptic graphics of its predecessor.

There's also a very handy new Frequency Profile menu that allows you to quickly change 15 EQ/damping parameters

simultaneously, additional stereo modes, and a new Cascade control offering increased flexibility in the serial/parallel routing of the early and late reflection engines, while quite a few parameters offer more extreme settings (for instance, the maximum rate of the reverb tail modulation can now be increased as far as 100Hz, for some almost FM-like sound effects). With such creative options, Aether has moved beyond the realms of 'reverb' and can now be almost considered an instrument: indeed, if you set its decay time to infinite, you can generate continuous soundscapes from almost any source.

The browser window has also been overhauled, with clearer section labelling and the ability to view any or all preset folders individually or in any combination. Being able to restrict preset viewing to, for example, plates, drums and instruments is doubly useful now that two 150-preset expansion packs are available, since these boost the total to some 600 presets! The Integrity pack concentrates on adding more halls, rooms, plates and other more traditional musical treatments to the collection, and very nice they are too, while the Creativity pack employs more extreme parameter values, offering a stunning set of mystical and thematic reverbs, plus various granular and modulation effects.

However, for many users the most significant changes will be within the reverb engine itself, which offers various higher quality options. I was very impressed with the sound of version 1, but in version 1.5, critical code areas are now implemented with 64-bit double precision, while new real-time and offline options for oversampling, modulation, and interpolation provide yet more clarity and detail.

Of course, you rarely get anything worthwhile in life for free, and the new higher quality settings do take considerably more CPU power (many presets took 40 percent of my Core 2 Duo 2.4GHz CPU at the highest real-time quality settings), but the difference is in no way subtle — switching from »

UAD EP-34 Tape Echo

» 1x to 2x oversampling, for instance, gives a significant boost to clarity, while 4x edges things that little bit further, to my ears bringing Aether's audio quality close to anything on the market. I already considered Aether a damn fine reverb, but with version 1.5 it has attained true excellence — miss it at your peril!

Martin Walker

£ **$249.95 (free upgrade for existing users); expansion packs $24.95 each or $39.95 for both.**

W **www.2caudio.com**

Universal Audio EP34

Formats: UAD2

Universal Audio's EP34 Tape Echo plug-in runs only on their newer UAD2 cards, and models the sound and features of the 'oldie but goodie' Maestro Echoplex tape delay, combining the features of both its EP3 and EP4 incarnations and offering delay times of 80ms to 700ms. Unlike the multi-head Roland Space Echo, the much earlier Echoplex produced a single delay, with the usual feedback control adding repeats and a sliding record head to adjust the delay time. Its tape loop and circuitry introduced distortion, wow and flutter and high-end loss that gave the impression the repeats were receding into the distance, very much like some of the delays used on Pink Floyd's *Animals* album.

A Record Volume control allows the input stage to be overdriven, while the Repeats knob goes from single repeats through all the normal flavours of echo to dub-style feedback. You can adjust the delay time during performance to get a pitch-slurring effect, just as with the real thing, and there's basic treble and bass EQ for the delayed sound. So accurate is the emulation that it even reproduces the Echoplex's 'squelch' effect when using high levels of feedback with low-pitched sounds, heard as breaks in the self-oscillation pattern as the circuitry gasps for air!

As usual, Universal Audio's designers have added a few concessions to the modern era including tempo sync, 'input level select' and tape tension, these last affecting the tonality and speed-change characteristics of the delays. For use in an aux send loop, the Wet switch mutes the dry signal, while Echo Send On/Off mutes the signal sent to the delay section. You can automate this to add delay to specific phrases.

Unlike the RE201 Space Echo plug-in, which was developed with Roland's blessing, the EP34 Tape Echo is not associated with the Echoplex name, so should be considered as an 'inspired by' plug-in, at least as far »

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» as lawyers are concerned. Subjectively, though, other than a lack of excessive tape hiss, the EP34 really delivers that 60-year-old tape-echo vibe. The degradation of the repeats is more pronounced than on the RE201 plug-in, and in some ways is more musical because of it. The dub-style runaway feedback sounds spot-on, as do the weird pitch effects you get when varying the delay time during performance. Predictably, it works brilliantly on guitar, but also adds a vintage flavour to other instruments and to vocals. In short, if you like echo effects and you have a UAD2 card, this is a must-have plug-in. You get a time-limited demo when you update to the latest UAD software, so take it for a spin. *Paul White*

£ $199.

W www.uaudio.com

Native Instruments Traktor's 12

Formats: Mac & PC VST & RTAS, Mac AU

For the benefit of any non-boffin *Sound On Sound* readers (*are* there any?), Traktor's 12 is a performance effects suite and one of five new products recently launched by Native Instruments to complement their Komplete range of instruments and effects. As the name suggests Traktor's 12 has, er, 12 effects, all of which have been derived from the DJ effects available in Traktor Pro, and can be used within any DAW or music sequencer. Like the other four, however, Traktor's 12 is not a plug-in in its own right, but is designed to be used within either the free or the full version of Guitar Rig. For the purpose of this review I downloaded the free version. This was easy enough to do (although a sizeable download at 300MB), and within half an hour I was making sounds worthy of a main-room DJ, which I guess is the whole point.

Once you're set up correctly, operating the individual effects is a straightforward business, as they all follow the same principle. As far as the effects go, I have to say I developed a bit of a penchant for the Peak Filter, which partied up everything so massively that my neighbour came round with some friends assuming I must be having an illegal rave. In a nutshell, Peak Filter adds a peak to the original signal in a specific part of the frequency spectrum, giving you DJ-friendly control over the frequencies you want to hear at a level up to four times louder than the 'original' signal. This became an invaluable way of drowning out the same neighbours when they came over the following night to ask me to TURN IT DOWN. It is also cleverer than its counterpart in Ableton Live, as it keeps the overall levels constant, where standard filters' output varies. Good to know if you can't face compressing your tracks after spending the night making small box/big box moves with your hands.

I also really dug the Formant Filter, which imitates the sound of vowels by morphing three band-pass filters into each other. You can control the nature of this effect by adjusting the 'sharpness' of the vowel sound or the 'size' of the mouth. And you can use the Talk button to morph between the different vowel sounds and adjust the brightness of the sound. I became quite obsessed with trying to emulate the 'wuh uh uh woo' sound at the beginning of Bon Jovi's 'Living On A Prayer', and got pretty close — to the point that er, some Bon Jovi fans came round... er, assuming Bon Jovi were playing... OK, you get the picture.

Overall, most of these effects offer something for everyone: even the frankly terrifying Ring Modulator and Mullholland Drive both do what they do very convincingly. Traktor's 12 is a welcome piece of additional kit which, most importantly for me, is straightforward and a lot of fun to use. My only reservation is that it has once again illuminated how much more time I need to spend in the studio...if nothing else, to finish my electro version of 'Lady In Red'. *Sarah Bowden*

£ €69 including VAT.

W www.native-instruments.com

Overloud SpringAge

Formats: Mac & PC VST & RTAS, Mac AU

Isn't it funny how you never miss old technology until something supposedly better comes along and replaces it? When spring reverbs were the only game in town, we were constantly frustrated by their 'boingy' metallic sound, with only guitar players seeming to like them, but now that everything has gone digital, we actually miss the things!

While some types of reverb can be emulated entirely by means of convolution, spring reverbs react differently depending on the nature and intensity of the input signal, so Overloud have employed a hybrid approach combining

both convolution and algorithmic simulation. Their SpringAge plug-in offers three selectable spring models based on various real-world units. 'AQTX' has a classic spring sound, well suited to guitar and vocals, while the 'S201' model is tailored for keyboards and synths and 'Angel' presents a brighter tonality.

Whereas the spring units in guitar amps usually have only a level control, SpringAge is very adjustable, and includes a Drive control to add a bit of preamp warmth. You can adjust the spring tension to change dynamic response and tonality, as well as making it more or less splashy using the dedicated Boing control. There's also a two-band parametric EQ for tweaking the tone, plus a stereo mixing function to allow independent panning of wet and dry sounds. The decay time of a typical spring reverb is fixed, but SpringAge has an adjustable decay time, so it is very flexible.

"It really does sound like a spring reverb, in every respect other than going 'boing' when you kick it!"

Best of all, it really does sound like a spring reverb, in every respect other than going 'boing' when you kick it! The characteristic, splashy twang varies with signal level and source material attack, just as it should, and it was great to be able to adjust the amount of spring character to suit the source material. On guitar, you can get very close to that Peter Green tone, as heard on 'Black Magic Woman', but you can also get a shorter, tighter reverb if you need it. Springs can be very flattering on voice, where there are no sharp attacks to set the things twanging, and used sparingly, you can also coax some nice vintage drum sounds out of SpringAge. I preferred the AQTX setting for most applications, but the other two offer useful alternatives. I was also pleasantly surprised by the very modest CPU overhead. Given the low cost of this plug-in and its compatibility with Mac and Windows platforms supporting AU, RTAS and VST formats (32-bit and 64-bit), SpringAge will be very attractive to anyone wanting to add a retro vibe to their mixes, especially as you can authorise it on up to three computers using a simple online procedure. *Paul White*

£ £82.25 including VAT.
W www.overloud.com

Drawmer A2D2

A-D Converter

HUGH ROBJOHNS

With its independently configurable outputs, this A-D converter is a little different from most of its competitors...

The pro-audio world is awash with A-D converters across a wide range of prices and with an equally wide range of features. However, the British manufacturer Drawmer's A2D2 stands out from the crowd because of its unusually versatile feature set and design topology.

This two-channel, high-quality A-D converter is housed in a 1U rackmount case and accepts stereo line inputs via electronically balanced XLRs. It provides two independent sets of outputs, each with AES3, S/PDIF coaxial and Toslink optical ports, all being active all of the time. There is also a quartet of BNC sockets, providing one word clock input and three buffered outputs. A fused IEC mains inlet feeds a linear power supply, which is factory-set for the local supply voltage (but is switchable internally), and completes the rear-panel facilities.

Overview

The front panel follows Drawmer's house style for its DMS products — a brushed aluminium facia, with a large blue illuminated logo at the centre. The right-hand side is dominated by a huge stereo bar-graph meter, which spans a 50dB range, with 1dB increments down to -10dBFS, 2dB steps onwards to -30dBFS and 5dB intervals for the rest. As standard, the meter shows green up to -7dBFS, with yellow to -1dBFS and red for the zero and overloading lights.

The left-hand side of the central logo carries the configuration controls, which start with a pair of rotary input-level controls scaled to accept peak levels from +28 down to -2dBu. An illuminated button switches between these controls and a fixed calibrated input level, set via a pair of front-panel, 24-turn level trimmers, with adjacent indicators identifying the current mode. The converter is aligned at the factory such that the calibrated mode conforms to the EBU specification, providing -18dBFS for a 0dBu input.

The sample rate of each of the two sets of outputs can be configured independently via two pairs of up-down buttons, each cycling through six standard sample rates from 44.1 to 192kHz, plus the external clock input — and the current mode is shown on two rows of LEDs. A further pair of red LEDs illuminates when the system has locked to an external clock. The final two buttons configure the two output sets to provide 24 bits of properly re-dithered, 16-bit output formats, the latter suiting legacy-format recorders.

Internally, the analogue front end is constructed from very high-quality Burr Brown OP275 op-amps. These feed a Cirrus CS5381 A-D converter running from its own crystal clock, operating at a fixed sample-rate of 210.9kHz. The digital outputs are delivered via Burr Brown SRC4192 sample-rate converters (which also provide the word-length reduction function) and Cirrus CS8406 transmitters. The output sample rate clocks for these two SRCs are derived from two separate, temperature-compensated AES Grade 1 crystal clocks generating the six standard rates.

This unusual arrangement (which is similar to that employed by Benchmark's ADC1) ensures that the A-D conversion is always running optimally and extremely stably, regardless of the clock source or rate, with the sample-rate converters handling the number crunching for the required outputs and isolating any external clock jitter neatly in the process. As a result, the A2D2 can be used to provide a high-quality 24-bit/96kHz output to feed a high resolution recorder, while simultaneously also feeding a 16/48

signal to a video system, or 16/44.1 format to a real-time CD-R recorder.

In Use

Hooking the A2D2 into my system was very straightforward, the calibrated alignment matching my own house reference levels perfectly. I have always found the quality of Drawmer's clock circuitry to be excellent, and the A2D2 maintains that standard with ease. The provision of three buffered Word Clock outputs also enables the converter to serve as the clock master for a modest digital system, if required.

The AES17 dynamic range test returned a figure of 100dB, which is below the state of the art, but still credit-worthy, and the THD+N noise figure was better than 110dB for normal signal levels. Although there are quieter converters on the market, in practice the A2D2 is more than quiet enough for real-world applications, and its versatility and configurability make up for the very minor limitations to dynamic range performance in many applications.

Careful listening tests revealed the A2D2 to be a fundamentally neutral, clean converter, with a strong bottom end and an open and airy top end, while the mid-range came across accurately and with masses of detail. Stereo imaging was precise, stable and spacious, with excellent width and depth. The excellent metering allowed levels to be monitored and controlled precisely, and the clocking flexibility accommodated every situation I could contrive with ease.

"I have always found the quality of Drawmer's clock circuitry to be excellent, and the A2D2 maintains that standard with ease."

One final point worthy of mention: a persistent issue I have with digital meters is the way most fail to indicate the normal operating range and safety headroom margin in an operationally useful way. The standard A2D2 does better than most in this respect, with the top six meter segments glowing yellow to warn of incursions into the headroom zone. However, with such a large and detailed scale, I felt there was a better option... and after discussing this with Drawmer I'm delighted to say that the company were very willing to supply a unit to my precise specifications. As a result, I bought a customised unit, which is now installed in my test reference system. The 'Robjohns Meter,' as I believe it is known in Wakefield, uses green LEDs from -50 up to -20dBFS, yellow from -18 to -9dBFS, and red all the way up to zero. This arrangement conforms with the broadcasting level standards (alignment level of -18dBFS and maximum permitted level of -9dBFS) and makes it very easy to see at a glance when signals are sitting in the 'normal' yellow region, and when they have crept into the (red) headroom margin.

Verdict

The A2D2 is a clever, versatile, and competitively priced A-D converter — and I'm impressed! ■■■

Alternatives

There are few A-D converters that match the sample-rate versatility of the A2D2, but the **Benchmark ADC1** is the obvious alternative, costing fractionally more than the Drawmer product.

Drawmer A2D2 £1146

PROS

- Very neutral and clean sound character, with excellent detail, clarity and stereo imaging.
- The immensely versatile outputs allow two entirely different sample rates and word lengths to be generated simultaneously.
- Switchable variable or calibrated input levels.
- Large and very readable metering (with optional Robjohns colour scheme!).

CONS

- Dynamic-range performance not quite up to state-of-the-art standards.

SUMMARY

An unusually versatile A-D converter with a clever internal structure that allows its two sets of independent outputs to be operated with entirely different sample rates, clock sources and word lengths.

£ £1146.63 including VAT.
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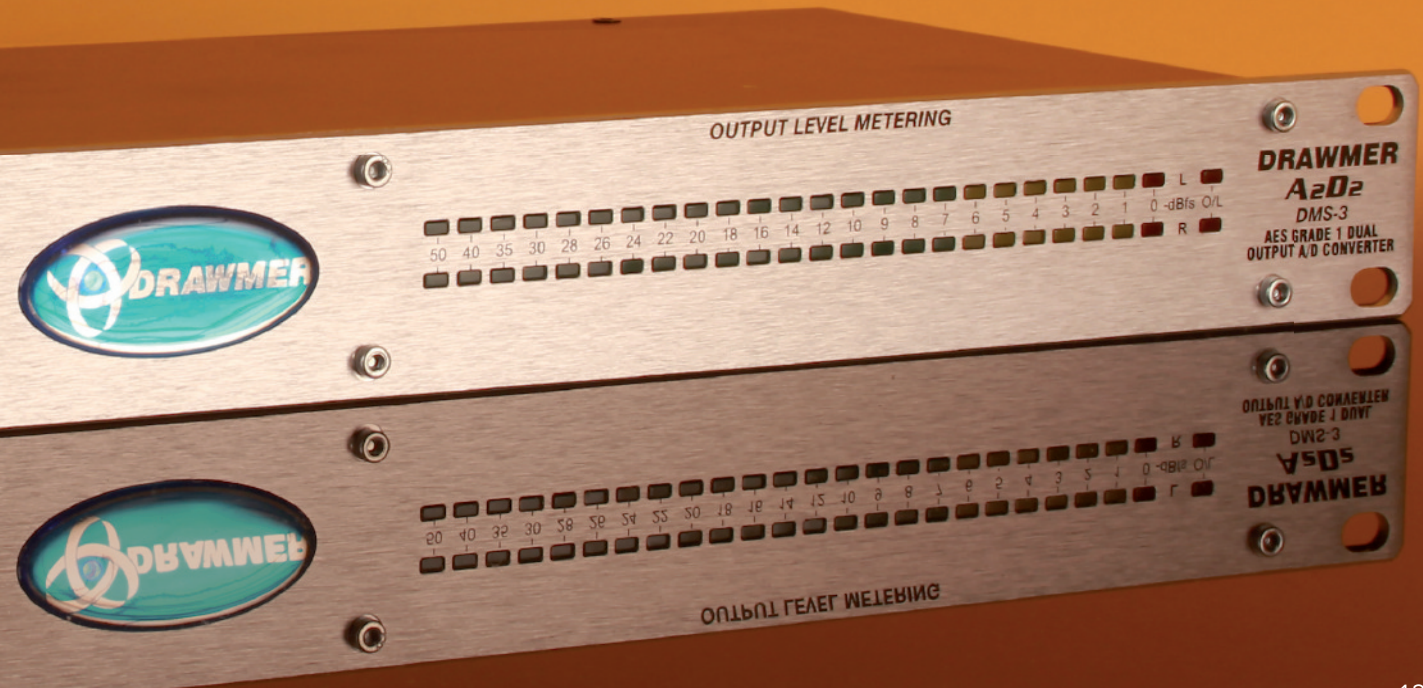


Sugar Bytes Guitarist

Virtual Guitar Software For Mac & PC

PAUL SELLARS

Electric guitar is one of the most difficult instruments to recreate in sample-based form. Sugar Bytes are the latest developers to brave the challenge.

Sampled guitars can be tricky. Sometimes they work well. Sometimes, even with the most painstakingly multisampled, velocity-switched libraries, they end up sounding a bit stiff and artificial. There have been previous attempts to tackle the problem by modelling the behaviour of guitars — and, more to the point, guitar players — in software, most notably Steinberg's Virtual Guitarist (reviewed in *SOS* December 2002 and September 2006). Sugar Bytes' Guitarist is in a similar vein, built upon large sets of guitar multisamples, with software mechanisms to mimic the interactions of pick, strings and fingers.

Get Plucked

Guitarist is available for Mac OS X (10.4 or higher) and Windows (XP, Vista or 7), and runs as a stand-alone application or a VST or AU plug-in. The installer is a 500MB download, and you're provided with a serial number that must be entered during installation.

Guitarist's graphical user interface is very pretty and well rendered, with its various moving parts all smoothly animated. The main window is divided into several sections. In the top left are pop-up menus for loading and browsing presets. In the centre of the display is a kind of piano-roll display for a built-in step sequencer. To the right are the effects and amp simulator. At

the bottom is the 'guitar' itself, on which mechanical 'fingers' move about forming chord shapes as the instrument is played (either via incoming MIDI notes, from Guitarist's internal sequencer, or both).

Presets are sub-divided into four different types: Global, Sound, Pattern and Chord. Sound presets store settings such as the guitar type and selected effects. Pattern presets store the strumming and picking patterns — right-hand activity, so to speak — while Chord presets store chord changes (left-hand activity). Selecting a Global preset automatically loads the matching presets in the other three categories. However, you can mix and match the different categories, combining the chord progression from one preset with the picking from another, and the effects settings from still another. User presets of all four types can be created and saved.

First impressions are positive. Several of the presets are impressively realistic, most are at least passable, and only one or two really clunk (the homage to 'Stairway To Heaven' is a bit painful). Guitarist is primarily a rhythm guitarist; although single-note melody lines can be programmed, that's not really what the instrument has been designed for, and they're likely to sound a bit mechanical and lifeless.

Strings & Stomp Boxes

Sample sets representing three different guitars are built in. Fender's famous Stratocaster and Telecaster are included, along with the rather less well-known Duesenberg Starplayer Special, a Les Paul-ish instrument with a fixed bridge and dual humbucker pickups. The Starplayer exists in 'Bright' and normal versions, while the Tele and the Strat have just one each (sampled from the bridge pickup in both cases, I would guess).

Extensive sets of fretted notes and open strings have been sampled, each one repeated several times. A 'round robin' function is used to alternate the playback of different samples in one of two ways. In Random mode, a sample is selected at random from the available pool for the note triggered. In Serial mode, the available samples for that note are triggered one after the other, before starting again from the top. Generally speaking, Random is the preferred option, since it better mimics the unpredictable variations of a real instrument performance. It's difficult to say, listening to notes in isolation, exactly how many different samples of each are provided, as the sample sets are stored in proprietary-looking files with a '.sbr' suffix, so I can't open them up and count — but I'd guess it was four or five at least. (The fact that I'm not sure is probably a good sign: in practice, the samples don't sound as if they're too often repeated.)

Guitarist's strings can be manually detuned, although only by a semitone in each direction, so it isn't possible (for example) to use the 'dropped D' tuning favoured by some guitarists, though the detuning may be useful if you need to match Guitarist's output to an existing take with slightly wayward tuning. You can also put the whole instrument slightly out of tune with itself, for a little extra garage-band realism. Double-clicking a tuning head returns the string to its proper pitch.

"The easy muting, damping and variable, choppy strumming functions can really bring a guitar part to life."

A good selection of guitar-oriented effects is built in. There are three different modules, followed by an amp simulator. The signal path is fixed and it isn't possible to patch the modules together in a different order. The effects are essentially insert effects, although there are Mix controls to adjust the wet/dry balance for 'FX 1' and 'FX 2'. FX 1 specialises in modulation effects, including a very nice Chorus, a good, throaty Phaser, and 'Phase Wah' and 'Auto Wah' effects, which are pleasant enough, if a little synth-y to my ears. FX 2 does basic delays and reverbs, which are fine, and an odd reverse-buffer delay effect, for the psychedelically inclined.

Next is a virtual wah-wah pedal. This is more sophisticated than the average guitar stomp-box, with an integral, variable-rate LFO to sweep the effect back and forth (this can be deactivated, and a MIDI controller assigned for manual control). The LFO can be set to retrigger with input signals above a given threshold (another way of producing 'auto-wah' effects), and Max and Min controls allow you to set the upper and lower boundaries of the effect's sweep. In use, it sounds pretty good, although perhaps a little polite, not being quite as nasal or nasty as the real thing.

Finally there's the amp simulator, which has three modes. The first offers Volume, Tone and Drive controls, and is cleaner than the second, which has Drive, Volume, Low, Mid and High controls and can get much fuzzier. Both are simple but effective, providing convincing reproductions of real guitar amplifiers. The third mode ('DI Box') has just a volume control, forgoing any amp simulation so you can use third-party effects instead. All of the effect parameters — and, indeed, most of Guitarist's GUI controls — can be controlled via MIDI (right-click a control to activate MIDI Learn).

One Thing After Another

To get down to the business of programming a guitar part, you need to turn to the pattern sequencer. This is the parchment-coloured tablature roll in the centre of the screen. There are five tabbed pages here, labelled Pattern, Animation, Song, Settings and About. The first three are where you enter and edit sequence data for patterns and songs. (The Settings page is where various instrument preferences can be set.)

The Patterns page looks a bit complicated at first glance, but is actually very easy to use. There are four main 'lanes' of data running left to right. The topmost lane carries 'trigger' data. For each step, there's a vertical bar with a handle that can be dragged up or down to set the amplitude for that step (all the way down is off). The lane below allows you to toggle individual strings on or off for each pattern step. The default chord voicings use all six strings, but here you can exclude strings as required.

The bottom-most lane is where you choose the chord to be played for each step. A root note can be selected by dragging up and down on the appropriate field. You can choose one of 12 different »

Sugar Bytes Guitarist
€199

PROS

- Quick and easy for a non-guitarist to use.
- Capable of convincingly realistic strumming, damping and so on.
- Useful built-in effects and amp simulation.

CONS

- The different guitar types are quite similar-sounding.

SUMMARY

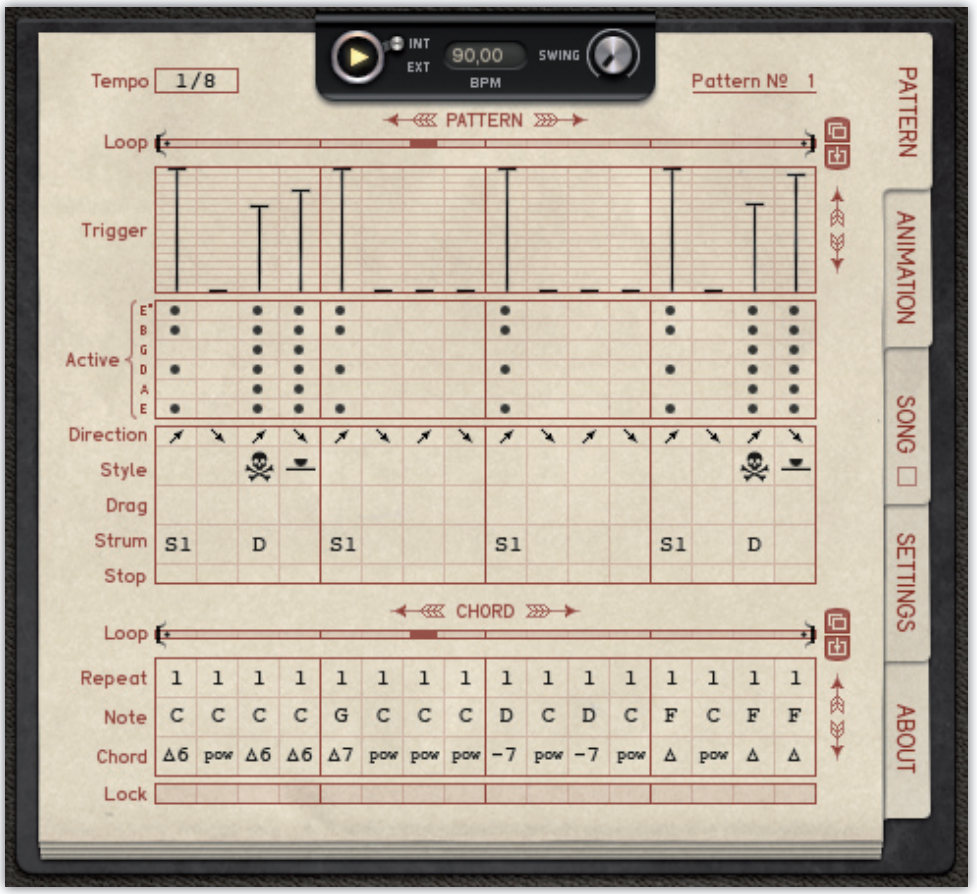
A clever and detailed emulation of the mechanics of (rhythm) guitar playing. Not foolproof, but capable of very believable results when used with care.

» chord types, such as 'major 6' or 'minor 7' from a pop-up menu, and each type is available in three different fingering variations. You can also create up to 24 user chords by clicking on the fretboard. These are assigned to the typewriter-style buttons below the guitar fretboard and can be triggered with MIDI notes or from the pattern sequencer. In addition to fixed chord shapes, the Animations page lets you program extra movements for each of the mechanical 'fingers' on the fretboard. A finger can be moved up or down by one, two or three semitones per step, to play (for example) a suspended fourth, or trills and ornaments.

The lane above allows you to choose what might be called 'articulation' effects for each step. These include the direction of the strum (up or down) and one of three Styles: the default is open strings, but alternatives are damping or muting with the heel of the right hand, and dead notes that are not fretted at all, with the strings muted with the left hand to produce a percussive noise. These are very effective, and with careful use of damping and dead notes it's possible to come up with choppy, funky rhythm guitar parts that don't sound at all programmed. A nice feature is the ability to use modulation wheel data to switch between these effects. For the first third of the wheel's travel, strings are open; the second third damps them; the final third produces dead notes. In this way, you can easily improvise and record variations in the phrasing of a guitar throughout a song.

You can set a step to Stop, which cleanly mutes the strings to stop a chord sustaining. You can also choose from one of several different 'Drag' effects, including vibrato, which sounds like a gentle wobbling of a guitar's tremolo arm, glissandos that slide up the fretboard to the target note, and smoother pitch 'glides', which can sound vaguely reminiscent of slide guitar. Another effect worth mentioning is the descriptively named 'Slow Down'. When triggered, this causes the rate of strumming to gradually decrease, so that the phrasing lazily winds down as it might at the end of a song. A simple effect, but convincingly guitarist-like, and a difficult thing to program in a conventional sequencer. Finally, you can choose from one of three slower strumming speeds (Guitarist's default strum is pretty quick, the notes of a chord sounding almost simultaneously), or choose a double strum, which fits two strums within the space of a single step.

The resolution of the pattern sequencer can be adjusted. Possible settings are quarter, eighth and 16th notes, with their triplet variations, plus 32nd notes. The pattern remains the same length when the resolution is changed, playback becoming faster or slower relative to the master tempo. When running as a stand-alone application, Guitarist provides its own internal tempo, and it will sync to a host application's tempo in the plug-in version. When you've programmed a few patterns (up to 24 can be stored), you can switch between them either by sending MIDI notes (patterns are assigned to keys on the on-screen pattern keyboard), or use Guitarist's own Song sequencer page to co-ordinate pattern changes. Sequencer patterns can be dragged and dropped onto the host sequencer, or to the desktop, and saved as MIDI files.



Guitarist's pattern sequencer provides the programming tools needed to make convincing use of its sample library.

Twang Etc

This all sounds a bit complicated, but it's actually very easy to work with. With half a dozen mouse clicks, you can quickly build up a pattern of changes that sounds a lot like something a guitarist might have played — and, more significantly, doesn't sound too much like what a keyboard player might have programmed.

Guitarist is cleverly thought-out and well-implemented. It's not perfect: the built-in effects are useful, but unremarkable, and its built-in sample sets are good, but perhaps a little samey. However, the easy muting, damping and variable, choppy strumming functions can really bring a guitar part to life. ■■■

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Test Spec

- Sugar Bytes Guitarist 1.01.
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Looking for a new audio computer?

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AVI Neutron Five

2.1 Monitor System

This interesting monitor system uses the natural roll-off of the satellite speakers to provide the crossover with the subwoofer.

PAUL WHITE

AVI are one of those small UK speaker companies who, like ATC and PMC, straddle the line between consumer and pro audio by building speakers that can fulfil multiple roles. Having already established a reputation for producing both speakers and amplifiers to a very high standard, it made perfect sense for them to develop their own three-way system suitable for home entertainment or studio monitoring.

To make it aesthetically appealing to domestic users, the company's Neutron system is reasonably compact, and is finished in flawless black piano-gloss with removable fabric grilles. Look inside, though, and you'll find some seriously good (and in the case of the sub, heavy!) drive units, meticulously designed crossovers and well-engineered amplifiers.

The Driving Range

AVI's drivers are now made by Sinar Baja in Indonesia, major world players in the loudspeaker business. They apparently acquired some personnel from the Vifa/Scanspeak/Peerless empire that had been the previous source for AVI drivers, and designer Martin Grindrod rates their drivers very highly, both on performance and consistency.

Conventional wisdom has it that there should be an electronic crossover between the tops and the sub, but Grindrod had other ideas when he designed the Neutron system. The sub's internal amp is fed from a low-pass filter, as you'd »

AVI Neutron Five 2.1
£1299

PROS
- Smooth, honest sound, with good bass extension.
- Compact and elegant.
- Includes digital inputs with remote-control switching.

CONS
- You're stuck if the dog eats the remote control!

SUMMARY
This little system seems well suited to both the consumer and small-studio marketplaces, and will be particularly attractive where one system needs to serve both purposes.

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PRIMACOUSTIC

The subwoofer is more complex than it looks!

» expect), but the tops are fed a full-range signal, with their natural low-frequency roll-off taking on the function of the usual electronic high-pass filter. This does limit the maximum SPL slightly, as low-frequency energy is still being sent to the satellites, but the benefit is that the sub integrates extremely naturally.

Handling the mids and highs in the 2.1 system I had in for review were a pair of AVI's Neutron Fives, the latest addition to the Neutron line. As the name suggests, the Neutron Five utilises a five-inch bass/mid driver, fitted with a curvilinear cone and driven by a 25mm voice coil. A 25mm soft-dome, doped-silk-composite tweeter handles the highs. The two drivers cross over at 3.4kHz via a passive, second-order Linkwitz Riley filter, built on a heavy circuit board with thick copper tracks, and using impressively large inductors and premium-grade metalised polypropylene capacitors. All this fits into a rear-ported cabinet built from 12mm MDF and measuring just 310 x 160 x 200mm. The inputs are on heavy brass binding posts and have centre sockets to accept banana plugs.

Power & Efficiency

Because the design is optimised for accuracy rather than loudness, the speaker efficiency is quoted as a modest 87.5dB (using the standard measurement of 1W at one metre), which though still quite respectable, means a reasonably powerful amplifier is needed to drive it effectively. The necessary amps reside in the sub, with 100W available for each of the two Neutrons, so that's not a problem.

While some companies keep their frequency-response figures a little vague, AVI are careful to present them correctly: the Neutron Fives measure flat ±2dB from 120Hz to 23kHz and are -6dB at 65Hz. This means that even without the sub the Neutrons provide an adequate low end for listening at lower SPLs in smaller rooms. Speaking of SPLs, my listening tests confirmed that whatever the maximum SPL might be on paper, it is more than enough for serious monitoring when the tops are set up as nearfield monitors.

Alternatives

This AVI system leans towards the tonal neutrality favoured by those engineering acoustic and classical music as well as pop, which makes it difficult to name a direct competitor at this price point. Models by companies with a similar philosophy towards tonal accuracy, such as PMC and ATC, tend to be rather more costly. If you're looking for a system of equivalent price, then check out Blue Sky's 2.1 systems or, at a slightly higher price point, some of the smaller Genelecs teamed with a sub.

The Sub Way

The Neutron Five sub looks pretty compact at 340 x 340 x 340mm, but as soon as you try to lift its not inconsiderable 26kg, you realise there's something pretty serious going on in there! The driver is a massively built 10-inch unit, and then there's the weight of three 100W analogue power amplifiers and their power supplies. The low-end response falls away gradually, but there's useful output down as far as 30Hz, which is impressive for such a compact unit.

While we are used to speakers with balanced XLR inputs, there are sectors of the hi-fi industry that believe all balancing circuits compromise the audio quality to some extent, and that at the short cable lengths involved, balancing offers no practical advantages. AVI have gone with gold-plated phono connectors for the analogue inputs (500mV sensitivity), as the humble phono provides a large area of metal-to-metal contact. Phonos are not designed to be plugged in and out at regular intervals — but once the speakers are installed, they're unlikely to be disturbed all that often.

There's more to the sub than just speaker and power amps, as the circular display window on the top right of the front baffle suggests. There are also digital inputs, and the sub has an inbuilt, very low-jitter D-A converter/receiver based on Wolfson chips, which are used in many high-end, stand-alone converters and are capable of extremely good results. The included infra-red remote control provides access to a number of inbuilt features that are useful both to the home-entertainment customer and to the studio owner (there are no physical controls on the speakers at all). The input to the converter can be chosen from one of four different optical ports, all capable of working at up to 24-bit, 96 kHz. There's also a USB input for direct connection to a music server, which can operate at 16-bit resolution. The remote control has dedicated buttons for selecting each of these inputs.

To avoid compromising the sound quality by using voltage-controlled amplifiers (VCAs), the remote volume control employs a stepped attenuator, adjusting the level in 0.5dB steps over a 100dB range. There's also a choice of 40Hz or 80Hz low-pass filtering for the sub crossover and a sub mute switch. Parameter values are shown while adjustments are being made, after which time the display reverts to showing which input is currently selected, this being remembered when the system is powered

down. AVI tell me that the sub is also available separately but is only designed for use with AVI's own two-way speakers, so anyone who already owns a set of Neutrons (or other AVI compact two-way passive speakers) could upgrade by adding this sub.

Performance

I already own a set of AVI speakers that I use for general music listening in my lounge, and although the Neutron Fives are somewhat smaller and less costly, they have a surprisingly similar family sound. Even without the subwoofer, they reproduce enough low end for low to medium-level domestic listening. In comparison with the vast majority of studio monitors, AVI speakers have a decidedly neutral tone, rather than the often artificially bright top-end and peaked up 80Hz bass exhibited by many monitor brands. This smoothness of tone makes them very easy to listen to, but detail still translates well. Those used to more brash monitors may take a while to get used

In addition to the two analogue inputs for the 2.1 system, the four digital inputs of the subwoofer allow you to select from different monitoring sources with the remote control.

to the sound, but when you're working in front of a pair of speakers for hours on end, the lack of sonic aggression achieved through low distortion is appreciated!

Adding the sub fills in the lower octave in a surprisingly unobtrusive way, adding the necessary depth without colouring the sound. My own preference was to use the 40Hz filter setting, as this seemed to balance well with the natural roll-off of the Neutron Five satellites. You can flip to the 80Hz setting for more impressive earthquakes if you happen to work in the same room as you watch your TV, and of course those selectable inputs means you can move from music computer to TV optical output to CD player (with optical or coaxial S/PDIF), without having to re-patch.

Opinion

Given the modest price, and the performance of which this system is capable, it has to be considered good value. The switchable digital inputs are a bonus for those who need them, and the high-gloss piano finish should look in keeping anywhere. The neutral sound may take some users by surprise, especially if they're used to monitoring on US-designed speakers with their typical 'forward' voicing, but they're comfortable to work with, more than adequately revealing of detail, and in even a modestly treated room, the stereo imaging is excellent. What's more, mixes that sound good on this system translate well to other playback systems. At the price they have to come highly recommended.

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STUDIO S.O.S

YOUR STUDIO PROBLEMS SOLVED

With over 50 Studio SOS visits under our belt, we thought it was time to discuss some of the most common studio problems we've tackled.

PAUL WHITE

Having now completed well over 50 Studio SOS visits, I thought this might be a good time to take a step back and consider some of the issues that we've encountered time after time in readers' studios. It really is quite simple to tackle many of the most common issues — or to recognise where there are insurmountable problems, so I really don't think we can ram home these points too much!

Of course, no two studios we've visited are the same, but one thing I have noticed is that the choice of equipment has only very rarely proved to be a major limiting factor when it comes to the quality of readers;' recordings and mixes. Much more often, the issues are about the speaker setup and acoustics, or how hardware and software is being used.

Your Room's Bass Response

By far the most common issue we're asked about is how to 'sort out the bass' in a studio — and problems with the bass end often show up in candidates for our Mix Rescue series. So let's start there...

Where a studio is set up in a typical domestic room or converted garage, you'll generally get a much more even bass-end response if your monitors face down the longest length of a rectangular room rather than across its width.

Small, square rooms (or worse still, cuboid ones) pose the biggest acoustical problems, especially if your listening position is close to the centre of the room: low frequencies tend to null out at that position, so you hear far less low end than your speakers are actually producing. The temptation is to EQ to compensate for the lack of bass and, consequently, you'll tend to produce very bass-heavy mixes.

The reason why square and cube-shaped rooms are so problematic is that all the peaks and troughs in the bass response (caused by reflections) occur at the same frequencies for each pair of parallel surfaces. In an ideal room, the lumps and bumps would be more evenly spaced. Proper bass trapping in a small square or cuboid room isn't really possible, as it would require more mass than you could realistically fit into the room!

To minimise problems with bass in small and medium-sized rooms, make sure your speakers aim down the length of the room, pointing towards the listener's ears. In this picture, you can also see the 'mirror-point' acoustic treatment that is used to absorb mid-range and high frequencies, resulting in better stereo imaging.

Tip

All you can do if you're lumbered with a small, square room is to ensure you keep away from the centre when making mix decisions.

The construction materials used in the room also play a part in its acoustics — solid brick or concrete walls reflect low-frequency energy back into the room, whereas plasterboard walls, lightweight doors and windows allow some low end to pass through, and they also absorb some sound energy through frictional losses. The more low-frequency energy that's reflected back into the room, the more likely you are to experience problems at the bass end.

To see how your room behaves, try playing back a chromatic scale of sine-wave bass notes (it's easy enough to create this using your DAW), and then check from your mixing position to see if some notes seem obviously louder or quieter than the notes either side. The more even they sound, the better. Sometimes making small changes to the monitor positions will help smooth out these problems. »

Tip

Avoid putting speakers in corners, as the bass end will become very uneven. Also avoid placing rear-ported speakers right up against a wall, as the wall reflections will otherwise tend to augment the bass.

Monitor Choice & Location

Something we see time and time again is people concluding that to hear the bass better, they need to use speakers with greater bass extension than a typical small nearfield monitor. Actually, in smaller rooms, where the low end is difficult to control, choosing large monitors with an extended low end — or systems that incorporate a separate subwoofer — will tend to make the problem worse.

A cable tester is a cheap but useful tool that can help you identify phase problems caused by faulty or badly made cables — and can save you a lot more headaches besides!

Tip

Pick monitors that are appropriate to the size of your room.

It is also important to place the monitors in the correct place relative to the mixing position. Ideally, the speakers should be set up symmetrically to form an equilateral triangle with the listening position, and with the tweeters pointing at or just behind the engineer's head. There should be nothing in front of the monitors that can reflect the sound, including excessively large areas of empty desk, and where the monitors are not mounted on rigid stands, putting them on good isolation pads makes a big difference.

Tip

You can improve the performance of simple foam speaker pads by placing one or two thick floor tiles on top of each, with a layer of non-slip kitchen mat between each tile, and on top of the top one to grip the speaker cabinet. The extra mass helps reduce the amount of vibrational energy reaching the desk or shelf, and also keeps the speaker cabinet more stable. Expect to hear improved tightness at the bass end.

Phase The Music

Inevitably, we come across some systems where the monitor speakers are wired 'out-of-phase' — which means that one woofer is moving in while the other is moving out. This results in a loss of bass, a lack of focus in the sound-stage, and an uncomfortable feeling that I imagine is rather like having your ears sucked out by a sink plunger! The cause is usually crossed-over positive and negative cables in the case of passive monitors, or an incorrectly wired balanced cable in the case of active speakers.

Tip

Invest in a basic cable tester. They're inexpensive, can be used to test all kinds of cables, and will let you know if your balanced cables are wired with the hot and cold crossed over at one end.

Acoustics

Thorough acoustic treatment is a complex subject, as the aim (in a mixing and monitoring environment, at least) is to achieve a consistently low reverberation time across the whole audio spectrum, and dealing with the low-end problems described earlier requires bulky bass trapping that may not be practical, or even possible, in a small domestic studio room.

Our usual Studio SOS approach of using relatively thin acoustic foam tiles or mineral-wool traps is only effective at mid-range and high frequencies, but it can still make a massive difference to a previously untreated room. With suitable treatment, you can expect to hear a generally better-focused sound, with more pronounced left/right stereo imaging. The thicker the foam or mineral wool, the better it works lower down the audio spectrum, and leaving an air gap behind the absorber also improves its low-mid performance to some extent. Mineral wool should always be covered with a breathable fabric, such as cotton, to prevent fibres escaping, while allowing sound to pass.

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Although we managed to get a decent result in this studio, our job was made much harder by the fact that it had been entirely carpeted. That's a cardinal sin of acoustics, as all it does is absorb at the high end, without tackling the rest of the frequency spectrum.

» **Tip**

Black weed-control matting from the garden centre works fine as a covering for mineral-wool panels, and costs very little. For something with a little more colour, try cotton sheets. Tablecloths from your local supermarket can do the job nicely!

Panel absorbers are intended to control reflections — and the usual way to deploy panel absorbers is to use a mirror held against the walls, so that you can see a refection of the monitor speakers when seated in your usual mixing position: where you see a reflection is where you need an absorber.

In larger rooms, further absorbers on the side walls can help dry up the overall sound — but you shouldn't need to cover more than around 20 percent of the total surface area. In a small room, you'll probably end up with one panel just forward of either side of your listening position, maybe one on the ceiling above your knees, and one behind each speaker. If you have space, putting another between the speakers will help as well.

The rear wall, behind the listening position, often works best if used to diffuse (or, in layman's terms, scatter) rather than absorb sound. That's something that can easily be achieved using shelves of books, CDs or unused gear, and a studio couch usually helps too. If there's bare wall left at the back, think about hanging another absorber.

Thin materials, such as carpet, simply soak up all the top end, while leaving all the mid-range and bass frequencies running riot, and that results in a claustrophobic, boxy sound. If you've committed the ultimate sin and carpeted all your walls (I never cease to be amazed by how many people do!), you can redeem yourself by sticking areas of reflective material on top of around 50 percent of it — any thin wood, MDF or plastic material will work, even old CDs. You can also put your foam or mineral-wool absorbers on top.

Tip

In rented rooms, you can hang foam panels 'picture style' by gluing an old CD to the rear of the foam panel near the top and hanging it over a nail or map pin. Window reflections can also be tamed using half-open slatted blinds.

Vocal Recording

Boxy-sounding vocals are rarely down to having the wrong microphone or mic preamp, or any processing with EQ and compression. In our experience, the more likely cause is your recording environment.

While a good commercial acoustic screen, such as an SE Reflexion Filter, will reduce the amount of sound reaching the rear and sides of the mic, it's equally important to avoid reflections from the wall behind the singer — so a foam sheet, heavy 'moving' blankets or a thick polyester duvet (not a feather one, as all the feathers fall to the bottom!) hung behind the singer will make a huge difference.

A good ploy is to hang a duvet across one corner of the room, and then have the vocalist sing with their back to that corner. Add a Reflexion Filter behind the mic and you should actually get much cleaner results than you would from a small vocal booth, many of which sound boxy and confined.

Tip

Avoid close proximity to walls when recording vocals and other acoustic sources, but also avoid the exact centre of the room or the 'focal' point of a bay window.

Too Many Grounds

It's well known that certain wiring configurations can cause ground-loop hum, but you can minimise the problem by running all your audio gear from a single power point, using a distribution strip to fan out to the various bits of gear in a 'star' configuration.

If you use balanced cables where your gear supports balanced wiring, in the event of a ground-loop hum problem you can disconnect the cable screen at one end to break the loop. Don't try to fix the problem by removing mains grounds, though, as this can be *very* dangerous in the event of a fault.

Ground loops can also affect the reliability of digital transmissions, even though the hum isn't audible. On one Studio SOS visit, we had a situation where the studio owner »

If you want a drier vocal recording, you need to tackle reflections both in front and behind. In this picture, Paul White is setting up a mic with a Reflexion Filter in front, and some acoustic foam behind — although a polyester duvet would serve just as well, if you need something a little more temporary.

» was using different power sockets around the room to power his system, and every time the house boiler kicked in, his audio interface dropped out! Moving to a star wiring system fed from one wall socket cured the problem. You may find that those occasional and otherwise inexplicable digital clicks and pops have the same cause.

Tip

Buy a mains socket tester to check that you have no switched live and neutral connections in your house wiring or, more seriously, missing grounds. We once did a show at the old Wembley complex before it was redeveloped, and found more than half the power sockets in the room we'd been allocated had wiring faults!

Too Few Grounds

We've come across a couple of laptop-based systems with severe hum problems caused by the system not being grounded at all. Electric guitar recording via a POD-style preamp is likely to really show up this type of problem: if the laptop and its ancillary gear runs from power adaptors, the chances are that these will have no ground connection, so you could end up with a mix of components without a single ground connection between them.

The answer is to ground the metal chassis of one part of the system, such as the audio interface, or the metal case on a jack or XLR plug, to the electrical mains ground. You can do this using a mains plug with just a ground wire connected, but it is advisable to insulate the live and neutral terminals inside the plug so that the ground wire can't come into contact with these by accident, should the cable get pulled from the plug.

Ergonomics

Many Studio SOS visits involve moving gear around to make it easier to use. In today's studio, where you may need to have computer monitors, monitor speakers, hardware controllers and music keyboards close at hand, good planning is essential to avoid frustration.

Having a small music keyboard that slides out from under your desk for general programming makes sense, even if you have a full-size keyboard off to one side for more serious playing. You might also find that one of the smaller hardware controllers that gives you hands-on transport controls plus a single motorized fader is adequate for your needs. If you can find a music keyboard with an adequate controller section that works with your DAW, so much the better.

If you're using lots of analogue gear, it's also worth planning which cables you need, and what cable lengths are required. We've seen so many studios that are just swamped by unnecessary metres of cable running all over the floor — and that's not the nicest of environments to work in!

Tip

To keep your cables short and tidy, it helps to be able to make your own. Buy a small soldering iron suitable for electrical work and learn to use it. It isn't difficult and it will save you money.

Muddy Mixes

We routinely listen to the readers' mixes when we visit their studio, and more often than not, 'difficult' mixes can be traced back to problems with the source material — often due to recordings made in an acoustically unsympathetic space, or failure to pay attention to mic positioning.

You may also find that some tracks in your mix have excessive low end that needs to be rolled off to keep things sounding clean.

Plug-in presets *can* be a great place to start... but if the signal doesn't rise above a compressor's threshold, it won't be doing any compression!

Learning how to wield a soldering iron can save you time and money.

Even instruments that produce no deep bass, such as acoustic guitar and voice, can generate low-frequency artifacts, and in the case of vocals, the lack of a pop shield (or the use of an ineffective one) can lead to air blasts producing very large amounts of low-frequency energy, even when no audible popping is evident.

A plug-in with a spectrum analyser display will help reveal this problem and a steep low-cut filter (10 or 24dB/octave) usually takes care of it. However, if vocal popping is so serious that it becomes audible or causes clipping, it can be very difficult to correct, so re-doing the vocals with a good pop shield might be the easiest option.

Tip

Check your recordings and mixes for excessive low-frequency energy — and reduce it. The result will be a much cleaner-sounding mix.

Plug-ins: Insert Or Send?

We've come across mixes with a CPU-hungry reverb in almost every channel. Even with modern multi-core machines, this can cause your CPU to trip up as the mix progresses. A more efficient way to work is to set up post-fade (aux) sends for the reverbs you need. A typical mix may only need two or three different reverb types, and by setting these up on sends rather than channel inserts, you can reduce the number of reverb plug-ins needed to three, and also blend them in any proportion on any channel.

Tip

Set up a default song template with the sends and suitable reverbs already in place. Also use screen sets if working on a laptop or with a single screen, as that will help you navigate your project with minimal fuss.

Presets

Nobody can be blamed for trying the presets that come with DAWs and third-party plug-ins, but some presets might not do anything without tweaks. We often encounter compressors that have been inserted on tracks but that are actually doing nothing at all — simply because the signal level never exceeds the preset's threshold! Although presets can be a useful way of finding the right type of compression, it's important to adjust the threshold while the track is actually playing, so that the gain-reduction meter shows the required amount of activity.

Although your ears should always make the final decision, gain reduction of 3-6dB on louder sections should be OK for most types of vocal and instrument recording. The same is true of limiters, gates, expanders, drum triggers — and any other processors that include a user-defined threshold setting: the best setting will *always* depend on the signal level of the audio you're processing.

Tip

Well-designed compressor presets can get you up and running very quickly with suitable attack and release times, but you must always adjust a compressor's threshold to suit the material you're working on.

Soundproofing

No discussion of our Studio SOS experiences would be complete without a word about soundproofing. People often don't understand the difference between soundproofing and acoustic treatment, so, here it is in short: acoustic treatment makes your room sound better, whereas soundproofing is designed to reduce sound leakage in and out of a space — and there's little or no overlap between the two.

When trying to reduce sound leakage, identify the weakest points first, such as doors and windows, because if they're not airtight, sound will leak through or around them easily. More mass equals more isolation, but you have to double mass to get a 3dB reduction in sound leakage — so a more common ploy is to use two boundaries with an air space between. Windows, for example, can be double-glazed, doors can be doubled up, so that you have one on the outside of the wall and one on the inside, and walls can have an extra plasterboard 'skin' added, again with an air space between them. The wider the air gap, the better the low-frequency isolation.

Floors and ceilings are more problematic, and while we've built simple floating floors using board resting on high-density mineral wool, we've never yet tackled a ceiling. Often, it really is better to find a way of working that makes less noise in the first place — such as using headphones late at night, or either DI'ing guitars or miking low-wattage amplifiers.

Tip

If you want to prevent sound leaking in and out of your studio, pay particular attention to any gaps around the doors and windows.

Finally, I'm sure that many of the issues covered here will be the same ones you've faced in your own studios, but there are always new problems to tackle — so do keep those Studio SOS requests coming in, especially if you think you have a problem that may be a little out of the ordinary!

Audio-Technica AT4047 MP

Multi-pattern Condenser Microphone

Audio-Technica have added multiple polar patterns to one of their already successful designs, bringing increased versatility in the studio.

HUGH ROBJOHNS

Regular readers may remember that we took a look at Audio-Technica's AT4050 ST stereo condenser mic in the last issue of *SOS*, and this month I'm completing my look at the company's new AT4000-series mics by putting the AT4047 MP through its paces.

Overview

The AT4047 MP is a multi-pattern version of the AT4047 SV fixed-cardioid mic, which has acquired a strong reputation, predominantly as a vocal mic, although it works well on a wide variety of musical sources. The MP provides selectable omnidirectional, cardioid, and figure-of-eight polar patterns, but is otherwise much the same as the SV, and maintains the same kind of 'vintage' sound that makes its sibling so popular and which is intended to resemble the character of early FET microphones. This character is achieved in part through its transformer-coupled output and the tuning of its dual-diaphragm capsule.

The capsule is a true DC-polarised design, requiring a standard phantom power supply, from which it draws a modest 3.4mA. The total dynamic range is specified as 141dB, based on a maximum SPL (for one percent THD) of 155dB SPL (rising to 165 with the 10dB pad switched in) and a self-noise figure of 14dB(A). Sensitivity is given as 7.9mV/Pa, which is about 8dB less than its sibling. It's also 5dB noisier than the AT4047 SV, but the maximum SPL is 6dB higher.

Audio Technica's marketing machine claims that the AT4047 MP has exceptionally low self-noise — but the current state of the art for a large-diaphragm capsule is around 6dB(A), and www.microphone-data.com currently lists 265 mics with self-noise figures that are the same as or less than that for the AT4047 MP, including 16 other Audio-Technica mics. So I think we can take that claim with the proverbial pinch of salt: the AT4047 MP is certainly a quiet and clean-sounding mic, but not exceptionally so.

The AT4047 MP is quite a large beast (both larger and heavier than its sibling), measuring 188mm long and 53.4mm at the widest part of its waist. It weighs 524g, and roughly 35mm of the mic's total length is attributable to the XLR output connector, which extends as a stem below the body of the mic. The capsule is protected within an oversized, double-layer, woven-wire grille which has been designed to minimise coloration due to internal reflections. Three miniature, flush-mounted slide switches around the base of the mic body allow one of the three available polar patterns to be selected, and introduce a low-frequency roll-off (from 80Hz at 12dB per octave), and a 10dB attenuator.

The entire body and grille of the 4047MP have a matte-silver finish, which is continued on the supplied low-profile, elastic-corded, cats-cradle shockmount

Audio-Technica AT4047 MP £729

PROS

- User-selectable choice of three polar patterns.
- Supplied with shockmount.
- Well-shaped tonality that helps vocals to cut through in a mix.

CONS

- Self-noise is higher than the standard AT4047 SV model.

SUMMARY

A multi-pattern version of the popular AT4047 SV, this new microphone brings considerable versatility and flexibility to a well-liked microphone, with negligible compromises in sound quality.

Alternatives

There's no shortage of multi-pattern capacitor mics on the market: www.microphone.data.com lists over 160 with the same three polar-pattern options as the AT4047 MP. Of these, the **AKG C414B-XLS** is an obvious competitor (or the newer and almost identically performing **C414 XLS,** which uses an electret capsule), as is the **Shure KSM44**.

(AT449/SV). This has a US-standard, 5/8-inch threaded stand fixing, and a 3/8-inch European adaptor was included. Also supplied in the kit is a dust cover and a foam-lined vinyl carrying case. In use, the shockmount proved secure and effective, and installing the mic was straightforward.

Performance

The 4047MP's frequency response in cardioid mode is very similar to its sibling's throughout the mid range and high frequencies, being exceptionally flat between about 80Hz and 3kHz. A modest 2dB presence peak extends between three and 10kHz, and the response falls fairly rapidly above about 12kHz. At the low end there's a mild proximity-effect bloom, but this is well controlled and only becomes excessive with very close working. When switching to omni and figure-of-eight modes, the response gets a little duller or a little brighter, respectively, but always remains fundamentally clean and clear, and is never objectionable.

The figure-of-eight polar response exhibits deep side-nulls with an even tonality, which only dulls slightly as the source moves around the edges. The omni pattern tends to flatten slightly at high frequencies, but this is a common side effect of dual-diaphragm mics (caused by capsule shadowing), while the cardioid pattern starts to become slightly more hypercardioid at high frequencies — which is, again, a common side effect. Overall, the switchable patterns are accurate and reliable, and the versatility is a welcome advantage over the AT4047 SV model.

The claim of a vintage sound character is, perhaps, misleading. The transformer and FET impedance converter do lend the AT4047 MP a slightly less clinical character than some other AT 40-series mics, but it wouldn't bear comparison with most valve mics if you're looking for that kind of vintage sound. Nevertheless, it does have a pleasant tonality, with a very well-judged blend of clarity and presence at the top end, and a nicely weighted bottom end. Proximity effect can be used to help balance the sound nicely, too, but if anything that characteristic seems less pronounced than that of the AT4050, the stereo version of which I reviewed last month. On vocals, I rarely felt the need for additional EQ to help bring a voice through in a mix: the presence peak seemed to provide all the clarity and diction enhancement that was required. The AT4047 MP also worked very well on acoustic guitar, grand piano and percussion tracks.

Audio Examples

In the course of reviewing this mic, Hugh made some recordings to compare it with a similar well-known condenser model. You can hear the results for yourself on the *SOS* web site at www.soundonsound.com/sos/dec10/articles /at4047mpaudio.htm.

Verdict

Overall, this is a very versatile and good-sounding mic. The added flexibility of selectable polar patterns is offset slightly by the raised self-noise figures when compared to the AT4047 SV, but I think most people will find the compromise more than acceptable. The greater tolerance to high SPLs is also an advantage when using the AT4047 MP for close instrumental duties. All in all, it's worthy and versatile microphone to add to the collection. ■■■

£ £729 including VAT.
T Audio-Technica +44 (0)113 277 1441.
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Roland RD700NX

Digital Piano

Roland are hardly new to the digital piano game, and with years of experience to draw on, their new stage piano should be something rather special. Let's find out...

ROBIN BIGWOOD

Roland's RD series of stage pianos has a lineage that stretches back into the mists of time, and in recent years it's been headed up by various models taking the 'RD700' moniker. All of them have offered a feature set that looks to be predominantly suited to serious live use: an 88-note, weighted keyboard; an emphasis on acoustic and electric piano sounds, supplemented by a broader 'gigging' sound-set; MIDI controller facilities; and a fairly straightforward user interface that blends ease of setup and sound selection with some degree of editing flexibility.

The RD700GX, released in 2008, upped the ante in this competitive part of the synth/keyboard market, not least because of its notably fine 88-note keyboard, which had textured, ivory-feel keytops and a convincing 'notch' in the keystroke that emulated a good acoustic piano's escapement action. Now comes the NX, the subject of this review, which appears to supplement the GX in Roland's line-up rather than directly replacing it. It retains lots of the best aspects of the GX, improves on some of them, but unexpectedly loses the odd feature along the way too. So have Roland finally discovered the formula for the perfect stage piano, or is this just a variation on a theme?

RD700NX Basics

The RD700NX uses Roland's top-of-the-line PHA III 88-note weighted keyboard, which, like the RD700GX's PHA II, distinguishes itself from the competition with its textured, ivory-feel keytops and escapement action. It is undoubtedly a classy affair, with the front edges of the black notes beautifully sculpted and the escapement resistance only noticeable when playing very quietly, which is just as it should be. The white keys are more textured than the black keys, and they offer quite a bit more friction when you slide your fingers back and forth — unusual, but by no means a problem. Slightly disappointing, but by no means unexpected, is that there's no aftertouch.

Lurking in a panel to the left of the keyboard is the typical Roland 'bender', a combined left-right pitch-bend and push-forward modulation joystick. Like all benders before it, it's sprung in both directions, and, though perfectly intuitive, it can infuriate some users who are more used to conventional pitch and

Roland RD700NX
£2299

PROS

- Excellent, tweakable, state-of-the-art piano sounds.
- One of the very best 88-note keyboard actions currently available.
- A comprehensive sampled sound set covering orchestral and band instruments.
- Good real-time control facilities, with plenty of pedal inputs.

CONS

- Limited programmability compared to a workstation or synth.
- The limited provision of only four front-panel faders struggles to cope with all control duties.
- The sound set is no longer expandable with SRX cards.
- The complexities of patch and memory structure are far from obvious.

SUMMARY

It's not without its limitations, and it won't suit everyone, but Roland have pitched the RD700NX's feature set just about right to make it a desirable and formidable tool for any gigging keyboardist.

modulation wheels. Above the bender are two buttons that can be assigned to control duties including turning effects on and off, applying octave transpositions, and so on.

The main control panel is gently angled up towards the player and is centred around a white-on-blue backlit LCD screen, which has enough resolution to display little graphical backdrops to some sound-selection pages. To the left of it are controls for configuring various effects, sound layering and keyboard splits. On the right lie preset-selection buttons and a big value dial, surrounded by cursor and other buttons used for navigating the NX's editing pages.

Overall, the RD700NX is a seriously chunky bit of kit, with metal panels and thick plastic end cheeks. The rear panel slopes at a jaunty angle, which looks good and makes accessing its sockets significantly easier. It weighs 25kg (55lbs), which, when combined with its bulk and length, is on the limit of what some will be able to cope with single-handedly. If you're planning on putting this in a flight case, make sure it has wheels.

What about the internal organisation? The NX offers several distinct sound generation systems, comprising 'Supernatural' acoustic and electric pianos, VK-derived tonewheel organs, and more familiar 'PCM' multisample playback. Also tucked away inside the NX are various drum kits and a GM2 soundset to provide Standard MIDI File playback compatibility.

Although it's technically more than 16-part multitimbral, the RD700NX played live can produce four parts simultaneously. Roland terms these parts 'layers' and calls them Upper 1/2 and Lower 1/2, but in fact it's up to the user to decide how they're configured. From editing screens accessed via the central Layer Edit button you can create multi-layer sound stacks, or set up keyboard splits, zone overlaps, and even velocity switches between one layer and another. In an improvement over earlier RD700 models, there seems to be no restriction on what sounds can be loaded, so you can have multiple Supernatural sounds, or several different tonewheel organ registrations in different key zones, for example. The four layers also get dedicated hardware volume faders. As for effects, there are single global reverb and delay/chorus processors, as well as a compressor and EQ, and basic controls for all of these lie within easy reach of the left hand. If you're thinking shared effects sound a bit 1990s, worry not: each layer is also equipped with its own pair of specced-up multi-effects units (connected in series), and these offer phenomenal processing clout, with no less than 84 algorithms on offer covering EQs, dynamics, 'lo-fi' treatmens, filters and distortion, through to delays, chorus, reverbs, pitch shifting and more.

In practical use, the NX is ready to go in about six seconds, and basic sound selection can be done with a single prod of a finger, as buttons are marked with sound names, not unlike on home organs or a typical child's keyboard! If the three flagship acoustic and electric »

» pianos and 10 basic additional sound types don't satisfy your needs, you just spin the value wheel to explore other ready-made variations. Using this system, 300 factory programs can be quickly recalled, and there's also a 100-slot user bank for storing and recalling your own creations. Not all is quite what it seems, though: see the 'Memory Lapse?' box to get the full story on the NX's preset and memory organisation.

Inside The Sounds

Being a live-oriented, predominantly preset-based machine, the RD700NX doesn't offer you full synth-style editing scope. However, some tweaking is certainly possible.

With one of the Supernatural acoustic piano sounds selected, pressing the Tone Edit button brings up an editing screen that allows quick adjustment of a host of piano-specific parameters (and which almost certainly point to a hybrid sampled and modelled sound-generation system). You can choose from 24 piano variants, adjust lid height, damper noise, duplex scale resonance, string resonance, key off resonance and hammer noise. You can also tweak stereo width, 'Nuance' (some sort of subtle phase-related effect), 'Tone Character' (mellow to hard) and 'Sound Lift' (a sort of dynamic-range compression). A further screen allows access to complex editing of velocity response, microtuning variations, sympathetic resonance and EQ.

For the Supernatural electric pianos there's another set of parameters: 'Bar Angle', Pickup Distance, Bell/Thump Balance and Bell Character. Additional edit screens give you access to a multi-effects processor and dedicated keyboard-amp simulation that can include tremolo and auto-pan effects.

For all the other sounds, Tone Edit gives quick access to the associated layer's two multi-effects processors, and some general parameters, such as mono, legato and polyphonic modes, portamento and bend range. One more edit page offers adjustments to filter cutoff and resonance, plus attack, decay and release times (which sometimes seem to relate to the synth architecture's amplifier, and sometimes to its filter envelope). These are all presented as relative values, so you tinker with what the sound already offers, rather than taking control of those parameters absolutely.

What Does It Sound Like?

I can confidently say that the acoustic pianos are the best I've heard from Roland, and they bear comparison with any sampled or modelled systems currently available, whether hardware or software-based (and that includes Roland's own V-Piano, I might add). The Concert Piano sound is woody, dark and brooding when played quietly, in the way that many concert grands can be, but takes on a brilliant, steely authority when played harder. Dynamic response is superbly matched to the keyboard action, and the overall dynamic range is prodigious, with no perceptible switching of samples. The Studio Piano is brighter, more superficially appealing, and better suited to luxurious pop styles, but just as complex and playable. The Brilliant Piano is the most hyped of the three, certainly more aggressive in the tenor and bass, but still a million miles from the trashy 'House Piano' patch typical of multi-purpose workstation keyboards.

Overall, the piano sounds retain more movement and complexity in the decay phase than is usual for hardware instruments, although there can still be more than a hint of sterility when you scrutinise individual notes held for more than a few seconds. Sympathetic resonance is implemented well, generating interactions between individual notes and chords, and an expansive shimmer when the damper pedal is depressed. While I'm on the subject of pedalling, the NX works well with the supplied DP10 pedal to produce half-damped and half-pedalling effects, and generates a subtle damper 'scrape' noise on pedal-down. This is strangely metallic in quality and long in duration — I haven't heard a real piano do anything quite like it — but it didn't bother me when heard in context.

To get some perspective I compared the RD700NX to the multi-gigabyte libraries of Synthogy's Ivory II, and while there are obvious tonal differences between the pianos on offer, the Roland wasn't outclassed in the slightest, which is pretty impressive.

The RD700NX's electric pianos are also sophisticated and successful. The front-panel 'Tine E Piano' Rhodes is very appealing, has a hard tremolo/autopan, and never becomes too heavy and loutish. The variations on the Rhodes are good too — a driven Dyno and some swooshy Suitcases — though on the whole they're quite bright and full. It seems the NX is

At The Back...

Befitting its pro-level appearance and price tag, the RD700NX has good connectivity to the outside world. There are unbalanced quarter-inch and balanced XLR stereo audio outputs, and a headphone socket. A single five-pin MIDI In socket is supplemented with no fewer than three separately addressable MIDI Outs (of which one can also act as a MIDI Thru). There are three quarter-inch pedal inputs, one for the supplied DP10 damper and two for additional switch or continuous-type pedals. Alternatively, Roland sell a piano-style, three-pedal unit (the RPU3), which can give you traditional una corda, sostenuto and damper functions (amongst other things). Alongside a display contrast pot there are two USB sockets; one an 'A' type for connecting a memory stick, and the other a 'B' type for hooking the NX to your computer. Power comes in on a standard three-pin IEC socket, and the rocker power switch is protected from accidental knocks by a plastic surround.

Missing from the rear panel are the Roland SRX sound expansion card slots that were present on the RD700SX and GX. I'm guessing that this won't be a deal-breaker for most people — it's not as if there aren't plenty of sounds on offer already — but it could be a drawback if you particularly wanted to equip your RD700 with a broader range of world-music, synth or other specialist sounds. You might be better off with the GX, if that's the case.

The RD700NX with optional KSG8 stand and RPU3 pedal unit.

not naturally given to generating those thick, sometimes drab early Rhodes tones that can work really well to glue together a combo. You also have to delve into the menus to tease out the really delicate, bell-dominated tones that can be so effective for quiet accompanying. The Wurlitzer sounds are certainly dynamic, but some of the amped variants are of limited use, descending into a distorted mulch.

Other keyboard sounds are mixed. There are uprights pianos in name, but they simply appear be slightly mangled versions of the grands. I don't think they'd fool anyone, and Madness tribute bands need not apply. Similarly, the Honky-Tonks are hideous, detuned monstrosities that you could never use seriously. If you're into 'character' pianos, it's unlikely the NX will do it for you; try the Nord Piano 88 or specialist software libraries instead. On the other hand, the Clavinets are superb, and there's any number of harpsichords, bells and excellent mallet instruments.

As for the rest of the sounds on offer, many are very fine indeed. The Hammond organ patches make use of the built-in VK-derived tone generator, and sound weighty and authoritative, taking advantage of some excellent rotary-speaker effects. Sampled highlights include some lush strings, surprisingly believable acoustic guitars, really outstanding acoustic and synth basses, and some solo brass patches that are much better than you'd ever expect. The NX's range of preset synth sounds feels a touch limited, and tends towards brassy leads and D50-inspired, twinkly, new-age genres. But all the »

The RD700NX's effects and EQ controls.

usual suspects are there, so you'll be able to whack out 'Jump' and 'The Final Countdown' no trouble. In short, all bases are well covered for the gigging player: this is by no means only a piano.

The Bigger Picture

In a world where a cheap laptop can generate a whole universe of sound, and big-money workstations offer full blown sequencing, sampling and every kind of synthesis, married to a dedicated control surface, a product like the RD700NX can look a touch anaemic.

Why would you spend this much money on something that's apparently a glorified preset machine?

Well, it's certainly not without its flaws and frustrations. If the sound you need isn't amongst the 1000 on offer, you're stuck — especially now that you can't even turn to SRX cards to expand your options. The sound editing on offer is neither deep nor particularly hands-on, and it's inevitable that at some point, live or in the studio, some users will find

Memory Lapse?

As I mentioned elsewhere in this review, getting going with the RD700NX and selecting its basic sounds is very intuitive indeed. But in the spirit of Japanese synths the full truth about preset selection and memory structure emerges only slowly, obscured by terminology and an operation manual that seems deliberately economical with the facts in many places!

So here's the skinny. The RD700NX actually has 1000 tones, presets, whatever you like to call them. The front-panel buttons offer no way to select these directly. Rather, they recall what for other synths is often called a 'performance' — a snapshot of a more complex setup that could include multiple tones loaded into the four layer slots, along with effects and performance settings. Roland term this a 'Live Set'. You only get to explore the full range of 1000 tones when you go into the Layer Edit screens, then you can step through them one by one using the value wheel, or use the front-panel buttons to jump to the first tone of a corresponding instrumental category. So the moral of the story is this: there are more sound variations and types on offer than is suggested by the Preset bank of Live Sets provided.

Keeping up? You'll need to. If you adjust one of the 300 Live Sets in the 'Preset' bank, or create your own from scratch, you can store it to a 'User' bank, which has 100 slots. Entire User banks can then further be saved to some other mysterious part of the RD700NX's internal memory as a 'Live Set File', so you can create archives of your preferred sounds ready for later recall. How many of these can be stored? I don't know. The manual doesn't say and I stopped counting when I'd saved six Live Set Files. In any case, individual Live Set Files in the internal memory can be copied to an inserted USB memory stick, and vice versa, so you're unlikely to run out of storage space.

As if this weren't enough, an alternative recall system for Live Sets exists, called 'Favorites'. It allows you to store any Live Set to an individual front-panel button, ready for recall with a single touch. Favorites mode is selected by pressing the Preset and User Live Set buttons simultaneously, which then gives you quick access to four banks of 10 Live Sets.

Still awake? I will be testing you on all of this later...

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Alternatives

The RD700NX has relatively few direct competitors, combining, as it does, the broad flexibility of the workstation with the immediacy of a dedicated piano-only stage keyboard. The slightly cheaper **Yamaha CP5** comes closest, though it has fewer non-piano sounds, arguably less convincing acoustic pianos, and less sophisticated MIDI control features. Coming in at around half the price of the RD700NX the **Kurzweil SP3X** is correspondingly more basic in most areas, but its 88-note keyboard can claim aftertouch capability. The most expensive of the lot, the **Nord Stage EX** overflows with vibe, and boasts piano sounds at least as good as Roland's, and certainly more varied. It's also a much better synth and Hammond clone, but offers no ROMpler sounds whatsoever.

» themselves having to compromise, or turn to a supplementary synth or ROMpler that has a broader sound palette. It doesn't help that the NX's patch and memory structure can feel obscure and unhelpful, nor that in the early stages it's far too easy to completely wipe complex multi-layer Live Sets you're constructing with a single errant button press before you've had time to store them. This is down to the user interface and display combination, which hasn't evolved much, if at all, from the first RD700 models, and as a result is looking and feeling quite dated. The use of four non-motorised faders to control eight internal and MIDI layers, nine drawbars and 16 assignable real-time parameters is far from ideal, and looks miserly when you consider just how prevalent the touchscreen is nowadays. The faders don't even offer a 'pickup' mode; parameters only snap coarsely to new values when you move a fader to a new position.

However, what Roland have done with the RD700NX, like RD700s before it, is to pull off a balancing act between flexibility and real-world usability. They've always been superb gigging instruments: in a theatre pit, my now-sadly-departed RD700SX handled everything asked of it, and would let me find, tweak, name and store a decent sound before the other workstation keyboard players had finished navigating their way through their 15th sound bank. The NX offers more, and better, of everything. The keyboard action and sounds are great, as is the provision of two effects processors per layer. Live Set recall is quicker than before, and does not silence held or decaying tones. Pedal assignment is hugely flexible, and there's now even an option for a pedal tap to step through Live Sets one by one. The NX makes a fine master keyboard, and all sorts of little setup options — like being able to override EQ settings programmed into a Live Set, to help cope with problem PAs — are really valuable.

In the end, it comes down to this: for many keyboard players, the RD700NX will be the perfect tool for the job. Significantly more capable than the average stage piano, and yet still very much a hands-on, player's instrument, it'll slot effortlessly into a variety of different roles and styles. Yes, there are eccentricities, and maybe an RD700 of the future, with a big touchscreen and still greater flexibility, would be an even more enticing prospect. But that doesn't matter. By any standards, the NX is a serious, versatile stage keyboard with a lot going for it, and it's easily the best RD700 yet.

The RD700NX's preset buttons make patch selection quick and easy.

Added Value Features

As well as the fine keyboard action and broad sound set, the RD700NX offers a few features you might not expect to see, but which could add a lot of value for some users.

Accessed via a dedicated front stop/start button, a range of preset drum patterns can be triggered. The sounds are very good, and the stylistic range impressive, but as a whole the feature feels badly undercooked. If only Roland had spared a few more buttons to equip these drum patterns with easily-triggered intros, fills and endings, you'd have a seriously useful rehearsal tool on your hands. But there's nothing of the sort, and what's worse is that the start button is far too easy to nudge inadvertently, which could be catastrophic in a live situation. It's also very unclear how rhythm volume can be adjusted relative to other parts, and, annoyingly the rhythm and tempo often changes when y,ou call up a new Live Set. Disappointing all round.

Less contentious are song playback features. Plug in a USB stick (which needs to have been originally formatted by the NX, and can only be inserted or removed when it's turned off) and you can load Standard MIDI Files to an internal playback sequencer. Audio files in MP3 and WAV format can also be played, with a varispeed option of up to 25 percent faster or slower (the time-stretched quality is not bad at all). During playback, all normal functions stay active, so this is very much a 'play-along' feature, especially when you consider the karaoke-style centre-cancel function buried away in the menus.

Strangely, the onboard MIDI sequencing facilities can't be used to record your own performances. However, you can record what you do as audio, direct to a USB stick. The process is a little convoluted, but it's there.

Of most value, by far, are the dedicated MIDI controller functions. Lurking behind the four internal sound layers are four MIDI layers (a front panel toggle switches over the layer buttons-and sliders to work on these instead). It's not immediately obvious, but User bank Live Sets can store and recall information about both internal and MIDI layers and their associated keyzones, plus MIDI program changes, MIDI Out socket and channel assignments, and a whole lot more. One thing that's good to see is that the USB and five-pin MIDI work separately and concurrently — not always the case with many comparable keyboards.

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The TASCAM GB-10 is the ideal practice tool for musicians of any skill level to build their technique

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NASHVILLE GUITARS

Recording Today's Country Guitar Sounds

With country guitars, what you hear on the record is what was played in the studio. We asked Nashville's leading engineers how they capture those tones.

DAN DALEY

Chris Leuzinger comes with baggage. Lots of it. One of the handful of session guitarists who play on most of the major-label country music recordings that Nashville methodically churns out, when Leuzinger walks into one of Nashville's numerous recording studios, he'll confront the engineer with a Vox AC30, a vintage Fender Bassman and a Deluxe Reverb, as well as several cabinets with various speaker configurations and two huge pedalboards, not to mention his own Royer 121 ribbon microphone. What there isn't a lot of is time. Although a session now lasts most of the day, rather than the three hours that were *de rigueur* in the '60s and '70s, the aim is still to finish three basic tracks per session, and guitars are still the foundational instrument on those records.

"We have to work pretty fast and still not make every guitar sound the same as it did on the last record, so the engineers and the guitarists collaborate pretty closely," says Leuzinger. "But the important thing — and what sets Nashville apart — is that the sounds are real. The [*live*] drums often just get used to trigger samples, the keyboard parts are often MIDI'd and the sounds are replaced, but it's not that way with the guitars here. The engineers don't do a bunch of Pro Tools re-amping. It's just the guitar being real."

The engineers agree. "I guess what defines what we do down here is that the guitar players and the recording engineers are pretty much on the same page — we go for the real sounds," says Bob Bullock, who has engineered most of Nashville's fabled pickers on sessions for George Strait, Shania Twain and others. "We live in a virtual age and we use that technology, but you just don't see it used on guitars in country music. Digidesign make their Eleven — their virtual guitar rack — it sounds good, and stuff like that and Pods have their place. But we're still going for organic guitar sounds. Guitarists still have lots of choices like they do with sampled sounds, but the choices really revolve instead around real guitars and amplifiers, and how they approach the music."

Up Close & Personal

While much of what is coming out of Nashville's country labels sounds a lot like rock music (more on that later), there are a few distinctly Nashville approaches to capturing guitar sounds. Close-in

Photo: Robert Knight Archive / Redferns

'Triple threat' Vince Gill.

Engineer Steve Marcantonio, who has worked regularly with Gill, among other country stars.

Session guitarist Chris Leuzinger.

microphone placement is one of them. A Shure SM57 placed within a couple of inches of a speaker cone, generally straight on or slightly off-axis, is the common starting point for most engineers. "There's not a lot of distance miking going on country records. We're usually not going for the over-the-top stadium-guitar types of sounds, so you don't usually hear a lot of room ambience," says Bullock. "Country tends to look for the organic sounds and get close up on them."

Steve Marcantonio has recorded plenty of country guitar stars, including the 'triple threat' Vince Gill — so-called because he plays, sings and writes, all amazingly well. "Most of the sound comes from the musician and they're listening to the front of their amplifier very close by," says the New Yorker, who moved to Nashville 23 years ago and has worked regularly with producers including Tony Brown, Dann Huff, Mark Wright and Josh Leo, and who calls putting a U87 up for the guitar on the other side of the room "a luxury". With the microphone close in, the amount of presence and brightness can be controlled surprisingly precisely with very slight adjustments in proximity (anything from two or three inches to as much as a foot away) and angle (never more than 45 degree off the centre axis of the speaker); the closer and more straight-on the position, the brighter the sound.

Picking the best speaker in a 2x10, 2x12 or 4x12 cabinet is another subtle but important choice. "Some of the guitarists will mark their favourite speaker with a piece of tape on the grille cloth," says Marcantonio. "When you're miking it close-in, the sound of the studio isn't going to affect the sound of the amplifier, and you can recreate the tone consistently from studio to studio."

If the guitarist is using an open-backed cabinet, a Sennheiser MD421 is generally trotted out and placed on the rear, several inches back and with the polarity button in the console input channel reversed, if necessary.

Cut With Ribbons

While the SM57 remains the stalwart of amplifier miking in Nashville, the Royer 121 has achieved near-parity in the last few years. It's usually placed next to the 57 and used to add more low-end capture compared to the bright mid-range of the 57. Sometimes it can be placed up against another speaker in the cabinet but on the same axis. "That ribbon can take a lot of sound pressure, so you can put it right up against the grille," says Marcantonio.

What brought the close-in culture about, though, is the fact that Nashville remains an oasis of ensemble recording. A typical tracking session will see five or more musicians in the same room; guitar amps are either gobo-ed or placed in isolation booths, though it's not unusual to see their doors open a bit. Country also remains a mix-as-you-go type of proposition, so engineers expect to record any effects or processing that guitarists plug in. "There are times when you come back to a song a few weeks later and put up a mix and wish you could pull the delay or something back a bit, but the effects are part of [*the guitarists'*] sound," Marcantonio laments. "Fortunately, what you don't encounter on country guitar parts is a lot of processing. Sometimes it's as simple as spring reverb and nothing else."

Stomp boxes tend to be noisier than pro audio gear, so Marcantonio has honed his chops on Waves' X-Noise and X-Hum plug-ins. With the single-coil-pickups of Fender's Telecaster and Stratocaster used on so many sessions, positioning the player is important to minimise hum and buzz.

Nashville is a songwriter's town, and composers put a lot of effort into demos. It's not unusual for the musicians who play on those demos (and they are a caste unto themselves, waiting years for a chair to open on Nashville's 'A-Team' of session stars) to hear guitar motifs they came up with played a few months later when the record version comes on the radio — an event that engenders a combination of pride and annoyance. "It helps for the engineer to listen to the demo, too, because the guitar sounds might have been a big part of what made the song attractive to the artist or producer in the first place," says Marcantonio. "They may want to use a guitar part and the guitar sound."

The Shure SM57, close to the speaker cone, forms the almost universal cornerstone of Nashville guitar sounds. These days it is often augmented with a ribbon mic such as the RCA 77DX (left), or, more usually, a Royer 121 (right).

It's All About The Vocal

Jeff Balding, who has recorded records for dozens of country artists including Sara Evans, Brooks & Dunn and Little Big Town, is also a fan of the Royer 121/Shure SM57 combo used close up (between one and three inches from »

» the grille) on guitar amps, which give guitars what he calls a "nice and furry" timbre that helps keep them present in the mix without increasing their actual level. He'll run the microphones individually through Neve 1073 or 1064 preamps, although on rare occasions he combines them together via a 'Y' cord into a single preamp. However, in a nod to how close country records are getting to mainstream rock, he adds a Sennheiser MD409 or AKG C414 to the close-mic mix, as well as adding room mics. One setup used on a recent session was an AEA R88 ribbon microphone on a stand four to five feet back from the front of the amplifier, to catch a combination of room ambience and direct sound. "It definitely helps the guitar sound fit in with a roomier drum vibe," he says.

Guitars are big in Nashville...

Balding reminds us that on a country music recording, the vocal carries the song and the song carries the record; all else — including guitars — is subordinate to that. "The vocal is going to be full bandwidth, so everything else has to be set around that," he says, "and to ensure that the guitars don't compete with the vocal, it's important to keep the guitars away from key vocal frequencies."

Triple Threat

Like Vince Gill, Brad Paisley is the rare 'triple threat' in country music: the artist who shines as a vocalist, songwriter and musician. Paisley's Telecaster work has made him the new face of Nashville guitar sounds and has earned him accolades in the form of Grammy and CMA awards, and engineer Brian David Willis has been behind the board for Paisley since the beginning. Willis takes a relatively conventional approach but with a different set of microphones.

"Brad was playing through a vintage Vox AC30 — loud! He had used a Sennheiser 421 on his demos, and when I began working with him on his first album, I added an AKG 460 to get more thickness in the mids, and I stayed with that setup for the first few albums," he explains. "Brad never really liked the 57 as much as other people do on guitar amps."

On the earlier albums, Willis positioned the 421 halfway between the speaker cone and the outer ring, at about a 30-degree angle off centre axis. The 460 was placed at less of an angle but about as close in as the 421, and he always recorded with the EQ flat on both channels. "A quarter of an inch movement with these microphones this close is all you need to dramatically change the sound," he says. "The setup

A Nashville Secret: The High-Strung Guitar

The Grand Ole Opry didn't allow drum kits on stage until the late '50s, and even then they often had to play from behind the curtain. As a result, acoustic guitars were what kept the time for country music, even in the studio, a tradition that continues today. A specialised instrument configuration was developed for it: the Nashville high-strung guitar, on which the lowest three strings are replaced with higher-gauge strings that are tuned up an octave. You can restring a guitar or use the processing techniques pioneered by long-time Nashville engineer/producer and Grammy winner Bil VornDick. He applies a high-pass filter at 160Hz and filters out the sound below that. "That essentially removes the first octave of the guitar; the lowest three strings range from 82Hz to 164Hz," he explains. The usual technique is to to layer two acoustic guitars — one full range, the other high-strung — both in mono. "The classic mono approach is to place a [*Neumann*] KM84 or 54 aimed at the 12th fret [*perpendicular*] to the guitar and set back about six to eight inches, then stack a high-strung on the second pass."


Bil VornDick.

For stereo acoustics, VornDick often uses a pair of KM54s. "The one on the left is angled down from the 12th fret and angled in towards the soundhole, and the right microphone is angled from in front of the player's right shoulder and towards the soundhole, keeping the 3:1 distance ratio in mind at all times. It really adds a third dimension to the sound," says VornDick, who has used this technique often with guitarists such as Tony Rice, Bela Fleck and Doc Watson. Alternatively, for acoustic guitars that aren't necessarily going to be doubling the hi-hat eighth notes but will still be strummed, VornDick likes to aim a Shure SM57 at the sound hole from slightly above it and eight inches away, coupled with a Neumann U67 or a Microtech Gefell UMT70, both condensers, in figure-of-eight mode to create an M/S pickup. "This setup gives you a lot of bite to the sound, lots of mid-range, and is great when you need the acoustic guitar to be able to hold its own with the cymbals and the hi-hat," he says.

gets all the beef out of a Telecaster but doesn't lose the twang." Willis ran each mic through a Neve 1073 mic preamp, and augmented the close mics with a pair of Rode NT1s in an X-Y pattern, set back about five feet from the front of the amp and pointing downward at the concrete-surfaced floor of the 3x6-metre recording room.

Brad Paisley's distinctive Telecaster style has taken the country music scene by storm (not to mention the White House, where this shot was taken).

On subsequent albums, Willis switched first to a pair of Neumann/Gefell M582s for room mics, then to a pair of Neumann U48s — which, in turn, have been superseded by an AEA R88 stereo ribbon. The room microphones add depth, but can also affect the articulation of the 16th-notes that make up blazing Nashville solos. "If we lost a little articulation in a passage I might be tempted to move the microphone inwards towards the cone, but that would lose some of the roundness of the sound. Before doing that I'd ask Brad to play it again, which he's always up for," says Willis. "Moving the mic inward is like dialling in a high-pass filter to lose murkiness that's getting in the way of every note being clean and sharp. Sometimes that murkiness is the enemy, but often you need it to give some weight to the notes in higher registers. It's something you need to keep in mind with Telecasters. If you can get the guitarist to play it again, I'd prefer to do that first rather than take a chance on changing a great tone."

Other techniques for keeping it clean include using only custom-made guitar cables fashioned from Mogami #2792 wire, and keeping the signal path as clear as possible, minimising the number of stomp boxes used.

One thing about Nashville engineers is that they are obsessive note-takers, and Willis is no exception. "If you have to come back days or months later and punch into a part, the only way you're going to be able to do that seamlessly is if you know exactly where you placed exactly which microphones on which amps and which guitars were playing through them," he says, noting that Paisley's setup has become more complex, with a second cabinet close-miked with either a U67, AEA R84, Royer 121 or even a 57 (depending on the song) through a Universal Audio 2-610 preamp. On one stretch of days in the studio, Paisley flew in bespoke amplifier builder Tony Bruno, who was hand-wiring circuit boards on the fly as the sessions went along, with Willis tracking each nuanced change.

Every Nashville engineer has his guitar tricks, but they'll also agree that a great guitar sound starts with a great guitarist. "Without that," says Bob Bullock, "all the microphones in the world won't do you much good."

Tascam DR2D

Digital Recorder

Tascam's mid-range pocket-sized digital recorder, the DR2D, supersedes the earlier DR1 — but does it have enough tricks to compete in an ever-evolving market?

TOM FLINT

The DR2d and its limited-edition pearlescent sibling, the DR2dw, are very much a step on from the DR1 I reviewed in *SOS* July 2008. That was one of the first generation of affordable digital recorders designed to fit into a pocket and record high-quality audio to solid-state media. The DR1 had some great features, but it lacked a few important ones, the most obvious of which was an on-board speaker for checking audio recordings and files. Tascam have had time to evaluate the DR1's pros and cons relative to the competition, so my expectations of the DR2 were much higher.

New Features & Updates

The thing that really set the original DR1 apart was the iPod-style scroll wheel, which was great for whipping through menus or song files, just like a dedicated MP3 player, and Tascam have wisely retained this feature. The screen menus and a number of the buttons are also similar. The casing is about a centimetre shorter, however, and not quite as wide, which results in a much tidier, more refined design.

But perhaps the key selling point of the DR2 is its ability to record two signals simultaneously, using a 'Dual' recording mode in which an alternative 'safe copy' recording is made anywhere from -6 to -12db down from the original. This is potentially very useful, and is sure to generate some interest, but the two other 'Dual' settings are arguably more intriguing. The first makes it possible to mix a mic signal with the line input to create a single file, and the second captures them both as separate files, just like a multitrack recorder. In this respect, Tascam are taking on Zoom's H4N, which has a four-track recording mode. The DR2 also makes it possible to overdub an input signal onto a pre-recorded file, be it an MP3 or WAV. This latter feature is something the DR1 could do, but Tascam have slightly altered its implementation.

The new model includes the small speaker that its predecessor lacked, located on the underside, with its own on/off button. Similarly, the DR1 lacked an integral screw thread for stand mounting, relying instead on an adaptor kit, so Tascam have added a thread, placing it near the speaker's switch.

One really nice addition to the standard kit is a wireless remote with a seven-metre range, matching that of Yamaha's similarly-priced W24 recorder. The slimline control is powered by a CR2025 lithium battery and includes transport and input-level controls, plus a Marker button. Tascam are clearly thinking of bands recording themselves from the stage, because the remote sends its beam to a sensor mounted at the front of the recorder near the mics, where a second record indicator LED is visible. This light flashes in record-ready mode and stays red when recording. The DR2's record button lights in the same way, of course, and by using that as visual feedback, you could do

Tascam DR02d £309

PROS

- Recording modes allow overdubbing, multiple-source and dual-level recording.
- Well constructed and nicely designed.
- Data wheel speeds up many functions.
- Low handling noise.
- Good signal-to-noise performance.

CONS

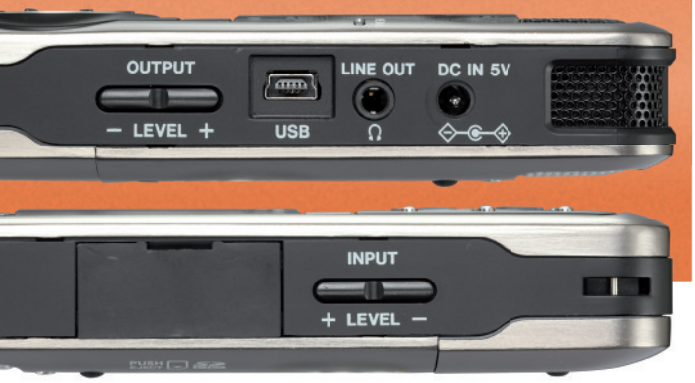
- Some of the setting up for dual and overdub recording is fiddly.
- No tuner, amp simulators or other guitarist-related features.
- No XLR or quarter-inch jack inputs.
- No X/Y mic setting.

SUMMARY

A significant step up from the DR1, this model is smaller and has many improved features. It lacks one or two good ideas tried out by the competition, but nevertheless has plenty of its own to offer.

Family Values

Tascam are very prolific when it comes to the manufacture and release of hardware recording products, and cater for very high-end studio and broadcast applications, as well as semi-pro home and live recording. In terms of portable, pocket-sized stereo recorders, their range started with the DR1, which hit the market together with a guitar-tailored sibling called the GTR1. At about the same time, they released the DR100, which was aimed at the professional end of the market, and came with a specification and price to match. In between these and the DR2d and DR2dw, came the DR7, which seemed to have a very similar feature set to that of the DR1, albeit with a simplified set of controls and a slightly more slimline case. The latest releases also include the DR8, which is a much more compact design than any of the above, and boasts extremely adjustable mics.



Most of the DR2D's controls are located on the two side panels.

remote operation from a front-of-house position too.

The new design dispenses with the DR1's bespoke battery, which could be recharged over USB: instead, the DR2 uses a pair of standard AA batteries. Although in the long run these will work out more expensive for the user, at least they're readily available and will be easy to replace in the field.

A further significant update is the addition of a 96kHz recording option (although note that there's no 88.2kHz setting); the DR1's maximum sample rate was 48kHz.

Casualties & Survivors

Not all of the alterations Tascam have made can be identified as definite improvements, however. The screen, for example, is marginally smaller than it was on the DR1, which obviously doesn't improve usability, although it does remain relatively large for this type of device, and still comfortably accommodates level meters, song position and file data, and general setup and system information.

The on-board mics of the DR1 could be incrementally turned 90 degrees to face vertically whereas those of the DR2 are fixed and can't be angled. While this might seem a retrogressive step, it is merely because the new setup is omnidirectional.

Other minor alterations include the shifting around of things like the hold button, DC power-adaptor input, USB socket and a few buttons, although these changes neither improve or hamper usability, in my opinion.

As for buttons, Tascam have removed the dedicated »

» input settings button, shifting its options into the menu button list. Also gone is the mix-balance control, used for changing the relative levels of a playback file and input source when overdubbing. This time around, the setting is done using arrow keys and an on-screen pop-up volume scale, but this is pretty straightforward.

Tascam have kept the six-option reverb processor, which can be inserted into the input path or applied to playback, even though it has also lost its button and has been relegated to the menu list. Retaining their button, however, are the playback control features, which allow the pitch of a file to be altered without affecting its speed (and vice versa), and also provide a means of cancelling out centrally panned material within certain user-definable ranges to leave the backing — all of which is designed to help soloists and guitarists who want to prepare practice material.

The remote control is a handy addition, and even makes it possible to control front-of-house recording from the stage!

The only input that's conspicuously absent is the quarter-inch jack socket, which was found on the edge of the DR1 nearest the operator, and was intended for non-powered mono inputs from dynamic microphones. However, the mini-jack mic input (supplying plug-in power), and line input, are both the same as before, and are found on the front edge between the on-board mics.

User Evaluation

Of the pocket-sized stereo recorders I've reviewed recently, the DR2 is one of the nicest to handle, feeling pretty solid and possessing a look that inspires confidence. In terms of size, styling and feel, there are similarities with Sony's PCM M10, which steered clear of fancy features but performed really well. Tascam have aimed for similar ground, but couldn't resist including one or two ideas taken from their multitrack recorders.

Control-wise, the functionality is fairly straightforward, although there are quite a few menus that make the setting up of things like overdub recording unnecessarily complicated. As for recording, background noise is far less prominent than that on Tascam's budget DR08. In fact, it is barely noticeable, even at the highest gain settings. Handling noise is also very low, and in that respect it performs far better than the DR1. The casing is made so well that it can't be made to creak when pressed hard, and the buttons are designed in such a way that they make very little noise for the mics to pick up. Nevertheless, Tascam must have done some clever engineering to stop eardrum-busting bumps and vibrations reaching the capsules in the way they seem to on many recorders.

Recordings made using the on-board mics sound very clear and detailed, if lacking in a little warmth as a result. The mics' omnidirectional polar patterns mean that the DR2 can be turned so that its screen is facing directly away from the subject and the result is the same as if it were facing the other way. Obviously, this is great for capturing ensemble performances where musicians are scattered around a room, particularly as the 3D imaging is very clear. On the down side, the mics can't be adjusted into an X/Y formation if a more focused stereo field is required.

"The new model includes the small speaker that its predecessor lacked, located on the underside, with its own on/off button."

Conclusion

In refining the DR1 to produce the DR2d, Tascam have both added and taken away. Fortunately, nothing too precious is missing: the quarter-inch jack input found on the DR1, which was useful for connecting dynamic mics, may be missed; but while guitarists might be disappointed that there aren't more specific facilities for them — such as a high-impedance jack input, tuner, or any signal processing — there is, at least, a metronome, and the playback control features are useful when jamming or practising.

Having 96kHz recording, the on-board speaker, a wireless remote control, a screw thread for stand mounting, and the ability to mix-record different inputs makes the DR2 a serious contender in the handheld-recorder market. Less obvious improvements, such as the significant reduction in handling noise, are also very important. The DR2 packs enough punch to compete with the best in its price range, and makes the decisions that prospective buyers have to make even harder.

Alternatives

Price-wise, the DR2 sits halfway between its main competitors, the **Yamaha W24** and **Zoom H4N**. On paper, at least, Zoom's feature list looks almost unassailable, with its adjustable mics, XLR inputs, physical modeling processor and 4-track mixer, but it lacks the remote facility of the other two (a wired one is an optional extra) and doesn't have a scroll wheel like the DR2. The W24 is the smallest and lightest of the three, but is not so solidly built. Other products at a similar price point include **Edirol's R09HR** and the **Olympus LS11**.

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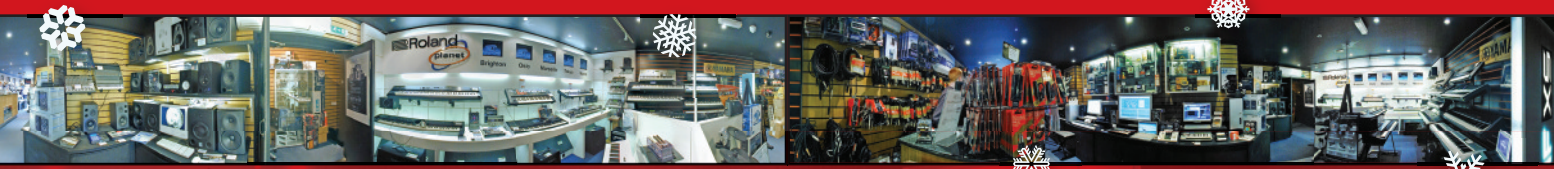

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Playback

Readers' Music Reviewed

Tolis Zavaliaris

II

For the first 30 seconds of *II*, you'd be forgiven for assuming that this was going to turn into yet another gaze up the rear end of an apparently under-rated schmaltzer. Not so, however: quickly progressing, as it does, from big, characterless chords to infectious guitar-led jazz, it's not often that the Playback vaults offer up something that is this pleasing in its originality.

Tolis Zavaliaris is a composer of some talent, taking catchy little jazz licks and fusing them with sounds of Persian inspiration, and it's this meeting of styles that works so well. Myriad percussive elements are also at play — most notably and successfully the tabla — and are given sufficient space to genuinely impress.

Though *II* has a rather amorphous overall feel, the album is an undeniable success and very enjoyable. The traditional Persian singing, in particular, is a pleasant surprise that underscores just what an epic Persian-jazz labour of love this record really is!

I dare say its downfall will be just how easy it is to listen to — it'll be coming soon to a dinner party near you. *Nell McLeod*

W www.toliszavaliaris.com

Tie Your Shoes To Your Knees And Pretend You're Small Like Us

Tie Your Shoes To Your Knees And Pretend You're Small Like Us

It's too early in the morning for this. I've had no breakfast, the coffee machine is broken, and a thousand multitracked voices are yelling their band name at me in a shit German accent. This is less music, more a bad acid trip in an underground film from 1968.

The feeling never quite goes away for the 28 minutes this album lasts. One minute it's muttered monologues over ukulele strumming, the next, experimental techniques for Jew's harp, backed with Bond-villain typing on a computer keyboard. Then, just when you're ready to write the entire thing off as a CIA-funded project designed to brutalise terrorist suspects, they unveil an entirely conventional and rather sweet Nick Drake-style instrumental for flute and acoustic guitar. It's original, brave, and probably unique. But I don't want to listen to it again. Not ever. *Sam Inglis*

W www.tieyourshoes.co.uk

Tony Alderman

A Single Word

They say that word processors have killed the art of handwriting, and until I turned to the accompanying letter, I was rather hoping this might be an album by the great Australian swing bowler Terry Alderman. Sadly not, and as if that wasn't disappointment enough, the letter begins "I made my CD on GarageBand." Eek.

Mercifully, though, generic Apple Loops are as notable by their absence as sandy moustaches and palpable LBW decisions. Instead, what we have here is an album of impassioned David Gray-esque acoustic songs, augmented by sparse organ or synth parts. From a production point of view, Tony's voice sounds very strange, as though he's somehow filtered out everything below 1kHz, and there are also tuning issues in the instrumentation on some of the tracks. He certainly gives it his all in the performance, with a tremulous vocal style that is earnest to the point of sounding self-parodic. I suppose that worked out all right for Antony & the Johnsons, so perhaps Tony should focus his energies on writing some slightly less conventional material. *Sam Inglis*

W www.myspace.com/tonyalderman

Dr Slaggleberry

The Slagg Factory

I did wonder, when opening Dr Slaggleberry's PR parcel this morning, whether their frankly awful name was nothing more than a stunt to get them written about. Obviously, if that were the case, it's an idea that is nothing short of genius, as here I am, drawn like a fly to the overly bright light of their electrical heavy metal trap... writing a review about them without even realising it.

"So what of Dr Slaggleberry?" I hear you ask. Well, you may be interested to know that the official Chambers English Dictionary definition of Slagg is "solid scum on molten metal". Need I say more? *Sarah Bowden*

W www.myspace.com/drslaggleberry

Silent Strike Lucian Ban & Alex Harding

3am

It doesn't make entirely clear whether Silent Strike is the name of the album or the band, but that small objection aside, this has perhaps the nicest packaging I've ever seen in a Playback submission. This is an act who have clearly thought as much about visual design as they have about their music, and it makes a very welcome change from the dodgy Microsoft fonts and dismal Photoshoppery that usually haunt the Playback shelves.

The music is equally accomplished, blending electronica and found sound with a variety of jazz and world-music instrumentation, all nicely played and recorded. Such fusions often veer into coffee-table territory, but there's enough of an edge here to keep the listener's attention. I'm not sure I'm quite ready for the other two albums they included, though — nice though they look... *Sam Inglis*

Clockworks

Everything In Colour

It's not often I'm sent unsigned music that reinvigorates my love of guitar bands, but I must admit that Clockworks' songs are almost good enough that I could be lured into a venue in Camden (and that's saying something) to check them out live sometime. Sound-wise, the band aren't a million miles away from Fall Out Boy, only less 'Rawk', and with considerably fewer tattoos or hormonal teenage boys for fans. If I had to be pushed on a description, I suppose I would say they were like Fall Out Boy crossed with Coldplay. But then I'm really running the gauntlet of lazy journalistic comparisons along the lines of "Daft Punk stuck in a lift with Chris de Burgh on drums and Tom Jones playing maracas on Dolly Parton's forehead," so really I should just stick to saying they are a more melancholy Fall Out Boy (if that was possible) and be done with it.

In seriousness, though, the songwriting and production on *Everything In Colour* are solid and mature, and on that basis alone, well worth checking out if you feel so inclined. A really promising band. *Sarah Bowden*

W www.myspace.com/clockworksonline

TnVVNuM

Ouroboros

TnVVNum is, and I am quoting exactly from their press release "short for Toele Nakku Vaadates Voib Naha Ukskoik Mida although we prefer to go by the abbreviation now which would be pronounced as separate letters... don't worry about the accents on the U and say it like 'You...'"

Yes, yes thank you Mr Tumnus(?) I Get It. You Can STOP NOW.

Their music is no less irritating than their over-eagerness to explain their name. Mr Tumnus sound like the Charlatans would if they had no singer, and keyboards played by a cat.

Obviously, this is something I'd pay to see, but other than Mr Tumnus's lion friend Aslan, I'm not sure anyone else would feel the same way. Perhaps they could focus a little more on their songs and a little less on the explanations in future. *Sarah Bowden*

W www.myspace.com/tnvvnum

Burn The Negative

In The Atmosphere

The wide open spaces of Burn The Negative's sleeve led me to expect epic indie guitar nonsense of a type all too frequently visited upon the unfortunate Playback reviewer. Surprising, then, that they actually sound like New Order, with a side order of goth. The material is decent enough, if considerably dated, and they coax some decent sounds out of their instruments, but the whole thing is let down by a nasty, crunchy edge to the whole mix, which makes it sound small and tiring on the ear. To my mind, this sort of thing needs to be glacial and pristine rather than harsh and grating. Oh, and there's the small matter of having a song about how cool Camden Town is. That would have been embarrassing 15 years ago. These days it's embarrassing *and* weird. *Sam Inglis*

W www.gung-horecordings.com

Beyond Reason

demo

I'm not sure this is a good thing, but what really stood out about Beyond Reason was their incredibly long and earnest biography, which read like a chapter of Motley Crûe's book *The Dirt* — had it all happened in a village near Bakewell, with Felicity Kendal fronting the band. Here is a little excerpt.

"...A chance encounter with drummer Scott led to a monumental first rehearsal with Taff, Keef, Scott and Nick 'Boycie Cotter' that lit the fires and made them realise that perhaps everything was in alignment. If only Keef wasn't going to become the father of twins in a few weeks' time, everything would have been perfect. So Keef's kids were born (little girls, in case you were wondering) and life (slowly) returned to normal. So at the beginning of the year Keef put the feelers out to the crew to see if they were still available and more importantly, if they still wanted it. And thankfully they did."

In contrast to all this Mrs Beeton recipe-swapping prose, Beyond Reason's songs are surprisingly angry. The final line of 'On With The Show' was so Kevin-The-Teenager-I-hate-you I had to play it three times to check I'd heard it correctly. A bit too pointlessly rebellious, given that the band seem to be averaging 40. *Sarah Bowden*

W www.beyondreasonband.co.uk

SAMPLE LIBRARIES

Original Music
Dance Drum Samples
Multi-format

Dance Drum Samples is a bargain-priced sample pack available to download from Original Music. It's a healthy 658MB in size and contains 116 WAV loops, of which 103 are presented in REX 2 format with accompanying MIDI files. The pack also includes 316 one-shot hits and 106 sound effects from which to create your own loops. Drum loops are organised into four folders according to tempo, which ranges from 70 to 175 bpm, the main stylistic focus being dubstep and drum & bass.

The first thing that struck me when going through the loops is that organisation could be better. Most of the loops have accompanying REX files, but some do not, which can be a little confusing and spoils the feeling of continuity between loops. It's also worth noting that there are some stray temporary files located in one of the REX folders. The other slight annoyance for me is the haphazard way in which the loops are labelled; for example in the 175bpm Drum & Bass folder, some of the loops are helpfully labelled 'Amen' (after the famous drum loop they were inspired by), but the majority are simply labelled with the letters 'A' to 'Z'. In future, I would suggest that the authors find more descriptive names for their loops.

The loops themselves are sorted into four folders: Drum & Bass, Dubstep, Processed Breaks and 135bpm. The Drum & Bass folder of loops is the pick of the bunch, and while not quite as good as those in more expensive packs, such as Big Fish Audio's Full Cycle, they are still very good. Certain loops from the Drum & Bass section, in particular, sound fantastic and could easily form the backbone of a composition. Also, having the loops as REX files extends their flexibility and I had great fun rearranging them in Logic. After the excellent Drum & Bass folder, the Processed folder was a little disappointing and lacking in variation, and the Dubstep folder, while good in content, contains only 16 loops.

The Dance Drum Sample pack also contains a generous 316 one-shot drum hits with which you can build your own drum kits. The samples themselves sound raw and unprocessed, rather than having a polished studio sheen, with some of the snare samples noticeably buzzing and ringing. They also demonstrate clearly audible hiss, which I found slightly offputting.

Overall, this release is a bit of a mixed bag. I really like the Drum & Bass loops and accompanying REX files, but find some of the other areas of the library less successful. If Original Music focused the pack more tightly around the Drum & Bass loops and made the Processed section more varied, the pack would be stronger, but even as it stands, the low price makes it reasonable value. *Geoff Smith*

Download £14.95 including VAT.

www.original-music.co.uk

Zero-G
Tonio: Male Classical Vocal
Vocaloid 2

I first saw Vocaloid demonstrated by Yamaha at the 2003 Frankfurt Musikmesse, and even though the chap presenting the demo spoke few words of English, the product's impressive capabilities were obvious. In a matter of minutes, the demonstrator had dragged a few notes onto a piano-roll display, tapped in some lyrics, added vibrato, dynamics and pitching characteristics and had the thing singing a song.

Tonio is the latest in a long line of products designed by third-party companies, using the Vocaloid engine to power their vocal sample libraries. To create Tonio, Zero-G recorded a classically trained male singer, apparently blessed with a register reaching from baritone to tenor. Nevertheless, opera singers around the world won't have to fear for their jobs just yet, as a considerable programming effort is required to turn each phrase into something that sounds realistic rather than robotic. While it's pretty easy to enter and edit notes and lyrics in the piano-roll style note grid, it's the process of sculpting those notes and words into something approximating a human performance that proves difficult. Vocaloid's tools are easy enough to understand, it's just learning when and where to use them that takes the time. Even the official Tonio demo, in which the voice is made to sing 'Una Furtiva Lagrima', is riddled with synthetic artifacts.

For most projects, it would simply be better to hire a local singer to do their stuff for half an hour — that is, unless something off-the-wall is required, or if the vocal lines are intended to be used as backing, where their shortcomings will be less noticeable. Indeed, Vocaloid allows the layering of up to 16 separate tracks, so complex — and relatively convincing — choral parts can be created.

But perhaps the real problem here is that something as precise as an operatic vocal isn't the ideal material for Vocaloid processing. We've all become accustomed to processed vocals in pop: in the '60s they were fed through Leslie speakers, the '70s saw the extensive use of vocoders and Eventide Harmonisers and, in more recent times, Antares Autotune has ruled supreme. But opera just sounds odd when coloured with an electronic effect.

Tonio works as a stand-alone product, which can be made to sync with a sequencer using Rewire if all the relevant settings are made in exactly the right order. Frustratingly, each time an edit is made to the Vocaloid project, the information has to be sent to the Rewire host before playback is resumed.

The DVD also installs a VSTi plug-in but, due to the bespoke MIDI data Vocaloid writes and reads, it doesn't integrate as seamlessly as a normal plug-in. Zero-G acknowledge that the plug-in doesn't work with Sonar, and I couldn't get FL Studio to see it either, but I did get the stand-alone component to sync with both DAWs (and Cubase) via Rewire, albeit with some inexplicable digital crackling in the audio output paths.

In summary, while the Vocaloid technology remains impressive, Tonio is definitely something to demo before buying. *Tom Flint*

£124.95 including VAT.

www.timespace.com

www.zero-g.co.uk

Big Fish Audio
Kontrol Freaks 2
Multi-format

This set of 40 hip-hop backing-track construction kits is the work of producer Soul-G and includes Apple Loops, Acidised WAV and REX formats. Each kit is showcased via a short demonstration full mix and then all the constituent layers are separated out as individual loops. The drums are presented not only as a mixed file, but also as a series of single-instrument loop and one-shots. This is handy, because the mixed loops are all digitally clipped — not any more than is common in a lot of hip-hop releases these days, to be fair, but I'd much rather make up my own mind about how much to clip my drums.

Overall, the groove is more about about lumbering impetus than nodding danceability, and the murderous seriousness of much of the emotional tone reminds me a lot of Notorious BIG in his prime: it might sound laid back, but it'll shoot you as soon as look at you! The nice tight kicks are a highlight, while snares are usually more understated, focusing the listener's attention on the muscly low-end and any vocal overdubs you might add. Basses are typically warm and round, with a hint of funk to them, frequently tipping the hat to Dr Dre.

There are few holds barred in terms of forces for the rest of the arrangement: you get a wide range of acoustic, electric and electronic instruments. However, it has to be said that the collection seems to lean more towards the latter two categories, and there's some tremendously characteristic lead and rhythm synth programming, in particular, with deceptively simple lines that somehow add up to more than the sum of their parts. The use of contrast is superb, with each sound differentiating itself beautifully from the rest, not least via artful effects use. This heightens the drama of momentary textural changes and also makes things a doddle to mix!

Soul-G's use of vinyl samples is arresting too, showing an instinctive ear for pitching, editing, and otherwise sculpting, such that they fit into arrangement pockets where their interesting details can really shine out. That enviable hip-hop skill of mining commercial potential from the cheesiest and most banal sources is also very much in evidence. I usually break out in a rash at '80s-style FM synth sounds or mark-tree glissandi, but he somehow manages to get away with using both of these offenders in his mixes without me batting an eyelid. In a word: voodoo.

And that's not the only magic here. I love the way Soul-G incorporates gritty, noisy samples into the track while maintaining an 'expensive' sound, and it's great that, without hiding his influences, he always seems to add something unique to the brew. Overall, this is a welcome change from hip-hop libraries that seem intent on regurgitating partly digested photocopies of the latest chart sensations. Leave those titles for the jingle merchants and go to Kontrol Freaks 2 if you want a dose of the real deal. *Mike Senior*

£71.47 including VAT.
www.timespace.com
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Best Service
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Kontakt 3+

The duo behind Cinematique Instruments have been composing music for films and documentaries for 10 years and have more recently extended their talents to the sampling of rare and strange instruments. Initially selling these individually and collectively as Kontakt-compatible downloads from their own web site, they subsequently joined forces with Best Service to offer a DVD collection of several dozen instruments employing the bundled YellowTools engine. Both formats feature identical samples, and when I auditioned them side by side, the only slight differences I could hear were between effects such as EQ and reverb.

This is a most eclectic 1.6GB collection, covering arcana such as the strident Bowed Psaltery, a fan-blown Magnus Harmonica Organ, shimmering Autoharp, plucked Kantele zither, subdued, muted Baritone Ukulele Muted, and the twang of the Celtic Nylon Harp. There are also dark, ambient, filmscape slices, glitches and weird and wonderful incidental effects from the Experimental Box. I was particularly taken by the 14 instruments in the Downbeat Box, consisting of pads created from treated guitars and feedback, loops and beats created with Korg's MS20 and the theremin-like Outta Space.

However, don't run away with the idea that this collection is only for music students and avant-garde composers; on the keyboard side, there are small upright and Rhodes Mkl pianos, a glockenspiel and a mangled electronic piano through a variety of stomp boxes. There's also some wonderfully varied metal percussion recorded in a smithy, along with pots and pans, struck wine glasses and a comprehensive collection of more traditional percussion.

Most importantly, the instruments are all very playable and expressive, and explored in a wide variety of ways, with multiple velocity layers, round-robin variations and release samples. To give you an idea of their scope, the Bowed Psaltery is recorded in short, staccato and tremolo versions, including up-bow/down-bow strokes and mod-wheel control over dynamics, processed into long pad-like variations as a set of performance loops and effects. It is even available as a tempo-sync'ed arpeggiated version with a clutch of tweakable variations!

There are a few rough edges: it's great to have round-robin samples on individual notes for added realism, but they shouldn't jump about in the stereo field as they do in the case of the Celtic Harp, while the Bowed Psaltery exhibits the occasional truncated note that doesn't occur in the Kontakt original. However, these are minor points that scarcely detract from the whole. Overall, whichever format you choose, Cinematique Instruments is a unique and inspiring collection of beautifully recorded, yet unusual sounds that will add extra spice and flavour to your music. *Martin Walker*

£175 including VAT.
www.bestservice.de
www.cinematique-instruments.com

Hands On Cubase

JOHN WALDEN

Even a handful of knobs and sliders on a MIDI master keyboard are enough to give you hands-on control of your Cubase mix.

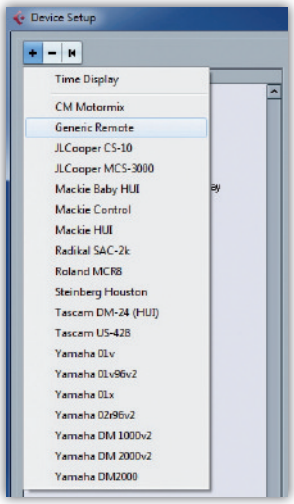
If you want to mix in Cubase using physical faders, but can't afford, or don't have the space for, a dedicated hardware controller, the faders and knobs built into many MIDI master keyboards offer a capable, low-cost alternative. In this article, I'll explain how to mix in Cubase using your master keyboard's controls.

For the purposes of illustration, let's assume we have a basic MIDI master keyboard that includes eight faders and eight rotary knobs. The most obvious way to exploit these controls for general mixing duties would be to assign them to volume and pan for a series of eight tracks in your Cubase project. So, assuming your keyboard is already connected to a suitable MIDI port for MIDI note entry, how do we configure these hardware controls for mixing duties in Cubase?

Remote Places

The first step is to get Cubase to recognise your keyboard as a remote control device, using the Remote Devices section of the Device Setup dialogue (accessed via the Devices menu). Clicking on the '+' button at the top left of the window opens a list of dedicated hardware controllers that Cubase supports, such as the Mackie Control. We require the 'Generic Remote', and once clicked, this is added as an item underneath the Remote Devices folder. When this is then selected, the centre-right section of the Device Setup dialogue changes to show a range of default settings.

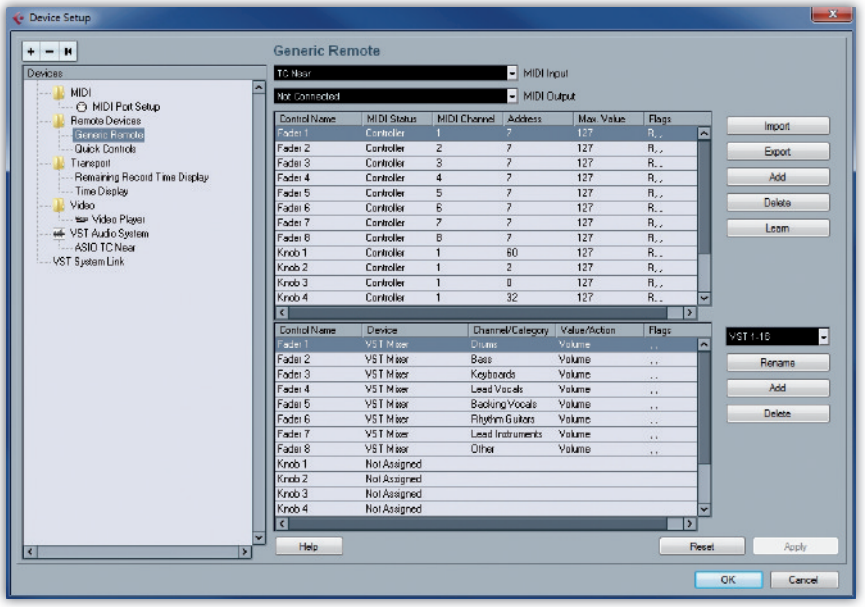
For most standard MIDI master keyboards, you'll need to specify the MIDI input (in my system, TC Near via my TC Electronic audio and MIDI interface) but can safely leave the MIDI output as 'not connected'. All this means is that Cubase will not be sending MIDI data back to the keyboard (which is not necessary unless your controller features motorised faders or some form of visual feedback). The bulk of the display is dominated by two tables. The upper table is used to define what controllers your hardware surface has. By default, it includes a combination of faders, knobs, mutes and sends. This is fully editable, and it makes sense to adjust the list so that it only contains entries suitable for the actual physical controls you have on your hardware surface — that's eight faders and eight knobs in this example. As you edit the upper table, the lower table changes to display the same list of controls.



If you just want to make use of some faders and knobs on your MIDI master keyboard, the Generic Remote is the controller you need to select.

The lower table is used to map a specific hardware control to a specific parameter in Cubase. This can be almost anything from a track fader to a particular effects parameter, but some care is needed, as these controls are 'fixed' to that single parameter until you return to the Device Setup dialogue to change them. The flexibility is considerable, but there is not much point is assigning controls to an effects processor that might not be used in your next project, so for a basic mixing setup the obvious candidates are volume and pan for some mixer channels. The question is *which* mixer channels?

If, as described in *SOS* March 2008, you tend to submix your various audio and VSTi tracks to a series of group channels, these make a good target. In this example, I have eight group channels in the project (drums, bass, keyboards, lead vocals, backing vocals, rhythms guitars, lead instruments and a catch-all called 'other'). As shown in the screenshot, clicking in the Device, Channel/Category, Value/Action or Flags column for any of



The upper table in the Generic Remote dialogue tells Cubase what controls your surface has, while the lower lets you assign these to mix parameters.

the controllers in the list produces a list of options that can be assigned. In this case, I've assigned each of the eight faders to control volume of one of the group channels in the VST Mixer.

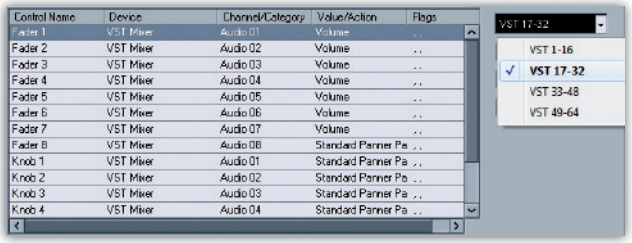
Learn Control

The second stage is to link each of your hardware controls to Cubase, using the Learn button, which makes the process very straightforward: simply select a control in the list contained in the upper table (for example, Fader 1), move the appropriate fader or knob on your hardware surface and then press the Learn button. The hardware controller is then linked to that virtual controller and, via the entry in the lower table, to a specific parameter in Cubase. You then simply repeat this process for each of the controls on your keyboard.

Bank Interest

While eight faders may be better than none, if you want to do more than mix a set of group channels, you'll need to use the four 'banks' of controls offered in the Generic Remote configuration. As shown in the screenshot above, the alternative banks can be accessed via the small drop-down list. In this example, I've configured the three other banks to provide volume and pan control for a series of 24 audio channels, but other targets, such as VSTi outputs, are possible.

Once this configuration is complete, if you open the (tiny!) Generic Remote window (from the Devices menu, as shown in the screenshot, right), you can toggle between the four banks as required – in this case giving us control over four banks of eight channels (the bank of eight group channels, and three banks of eight audio channels). Hey presto! You now have hands-on control of a key part of the mixing process. Combine this with recording your fader movements as automation data (a topic for a different column) and those complex mixes can suddenly seem to be much more manageable!

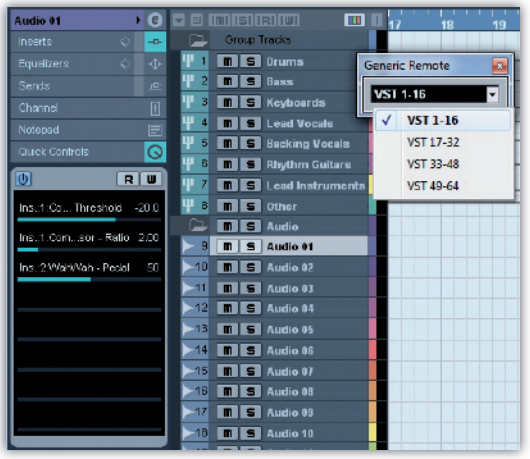


In the second 'bank', the eight faders and knobs have been assigned to volume and pan for the first eight audio tracks.

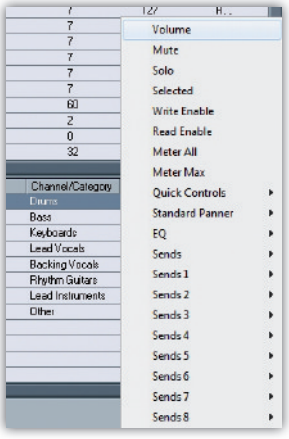
Project Translation

The Generic Remote settings are global in nature: once configured, the settings apply to every project. This has positives (you don't have to repeat the configuration process from scratch for each project) and negatives (not all your projects will require the same configuration). The approach also has a number of quirks, perhaps the most significant of which is that it is sensitive to the order in which tracks are placed in the Track List of the Project window.

Consider the example used above where the eight group channels happen to appear at the top of the Track List. If, for instance, an audio track were to be added above these group tracks, it would bump down all the hardware controller assignments by one. Fader 1 of bank 1 would now control this audio track, rather than the first of the group channels. At first, this can cause some confusion, but once you are aware what's happening, it's easy to work around the problem: it simply requires a consistent approach to ordering your tracks in the Track List. I often place group channels at the top, followed by audio tracks. The latter can be put into folders if required; it's the overall track order that's important.



The tiny Generic Remote window allows you to switch your hardware device to control multiple banks of parameters. As described in the main text, the Generic Remote configuration is sensitive to the Track List order. Here, the eight group channels are followed by 24 audio tracks. Also note the Quick Control on/off button in the Inspector pane — which is useful for disengaging the Quick Control system when required.



Clicking on any entry in the lower table (in this case, the Value/Action column for fader 1), brings up a list of the possible assignments.

Thankfully, the Generic Remote configuration options include import and export buttons. If you do need to configure your hardware controls in different ways for different project types, these can be saved and subsequently recalled, allowing you to develop any number of different configurations as required.

Generic Remote, Or Quick Controls?

Whereas a Generic Remote setup is most suited to mixing operations, the Quick Control system (*SOS* July 2009) is targeted at hardware control of individual tracks, with up to eight parameters in the currently selected track capable of being tweaked via a hardware controller. The two systems therefore complement each other very well.

One quirk is that it is possible to have both systems operational at the same time, and thus have the same hardware control linked to two different and unrelated parameters in Cubase. If you are not careful, you might find yourself inadvertently changing two parameters with one controller movement! Thankfully, the Quick Control system can easily be disabled via the Quick Control panel of the Inspector, allowing you to use your Generic Remote configuration. The most straightforward way to temporarily disable the Generic Remote system while you use Quick Control is to open the Generic Remote dialogue box from the Device Setup window, and then deactivate the MIDI input to the Generic Remote. Although this approach may be a little clumsy, it does at least do the job.

Looks Different

DP's latest version brings the ability to change the look and feel of the program. We find out how, and explore other additions in v7.2

From the neat and tidy 'MOtools' to the futuristic 'Plasma', DP7.2's Themes feature lets you choose a look that suits you.

ROBIN BIGWOOD

There's long been plenty of heated debate about the look and feel of DP, not least since its make-over back in the days of DP6. Users complained that the predominantly bright, light-grey windows are tough on the eye, especially at the end of long sessions into the small hours. A few complained that the whole environment was too boring! This led to a thriving interest and market in interface hacks, mostly centred around commercial 'mods' available from www.ampguimods.com. It all had a bit of an underworld vibe and was never likely to get the blessing of MOTU, but there were many happy customers who felt their DP experience was improved.

Clearly MOTU were well aware of all this, because in DP7.2 we have 'Themes' — essentially a built-in user-interface modification scheme that offers control over basic colour, as well as a way to swap out graphical elements of the interface. There's now a dedicated page for it in the Preferences window, and there you're offered a list of Theme presets. Click on one, then click the Choose button, and DP may be literally transformed before your eyes. You can make two simple modifications to each preset: to its Meter colour, and to the colour used for selected data. In each case, there are three straightforward, self-explanatory options.

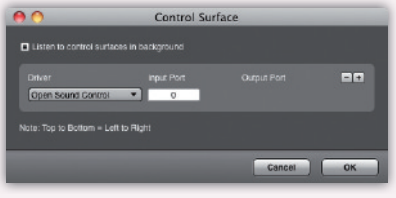
If you still can't find a result you're happy with, though, you can go further. First of all, you could use a third-party or custom theme, and designers such as www.ampguimods.com that I mentioned before now offer their own DP7.2 themes. To install these, click the Open [Themes Folder] button in the Preferences window, or go directly to /Library/Application Support/MOTU/Digital Performer/ Themes in your Mac's Finder, and place your new theme's package there. Restart DP and it should be available. However, if you're of a fearless, experimental inclination, and handy with an image-editing application, you could also try a bit of DIY.

In that same Themes folder, you can open up the themes packages (or, even better, duplicates of them that you've made, so you don't mess up the originals)

MOTU On The Phone

In July's DP workshop, I described how to set up DP7 to use the AC7 control-surface application for iOS devices. Soon after, MOTU went one better and released their own dedicated app, DP Control, which has the distinct advantage of being free, and much easier to use. To try it:

- **First download the app for your iPhone, iPod Touch or iPad, from the App Store. Search for it in iTunes, or use the link that's currently up at www.motu.com.**
- **Next, both your DP Mac and your iOS device must be connected to the same Wi-Fi network.**



- **In DP7, visit the Setup menu, and choose 'Control Surface Setup...'. Use the little '+' button to add a driver, and choose Open Sound Control from its pop-up menu. Then simply OK the dialogue box to dismiss it.**
- **On your iOS device, launch DP Control. It might all work beautifully, straight off. If not, go into DP Control's Settings view, and use the Change Connection button to set the link to your DP Mac.**

As soon as the connection is made, DP Control is a beautiful thing, letting you work from vocal booths or on remote monitoring systems. What's more, the mixer allows true multi-touch adjustment of faders and pans. A glimpse of the touchscreen future!

by right-clicking on them and choosing 'Show Package Contents'. Drill down into Contents/Resources and you'll discover that DP's interface elements are stored as PNG files, and it's pretty easy to modify or replace elements within them to really customise your own DP environment.

Range Rover

DP's automation is a wonderful thing, and the different operation modes available for it add to its flexibility. In DP7, there are four new ones — Range Touch, Range Latch, Range Trim Touch and Range Trim Latch — and, like the others, they're most visible and most easily accessed in the Mixing Board, where a different mode can be selected for each track. If you need a recap of how DP automation works, check out www.soundonsound.com/sos/jun05/articles/performernotes.htm.

What the new Range modes add is the rather useful possibility of resetting or adjusting automation values within a specified time range. That makes them a fast, easy, hands-on alternative to the fiddly process of deleting or modifying automation data graphically with the mouse, in the Sequence Editor.

Here's a typical scenario. You've got a heavily automated mix, and are happy with most of it, but you want to try out some completely different levels and pans for a group of tracks, in one small section of the song.

1. Make a time-range selection for the section and tracks you want to work on. A quick way is to simply drag in the time ruler, making a selection on all tracks.
2. In the Mixing Board, record-enable automation for the tracks you want to remix, and set their automation modes to Range Latch.
3. Start playback within or a little way ahead of the region specified in step 1.
4. Still in the Mixing Board, start setting the new levels and pan positions. As you do so, new, static automation values are written for the time range you selected, regardless of whether you're actually in it, and instantly replacing any similar automation that was there before.
5. If you want more attempts to really refine your mix, just place a Memory Cycle loop around the same section. A really quick way is to choose 'Set to Selection Bounds' in the small Memory Cycle pop-up menu next to the 'start' and 'stop' values in the Control Panel. Then start playback once more and keep tweaking until you're happy.

The difference between Range Latch and Range Touch is subtle. In Range Latch mode, automated faders or knobs stay still as soon as you begin to adjust them during an automation record pass. Data outside your selected time range won't be changed, but you can 'zone in' on your new settings in the run-up to it. In Range Touch mode, faders and knobs always track existing automation, other than when you're actively adjusting them, or when the playback wiper is within your selected range. Try both out to see which you like the best.

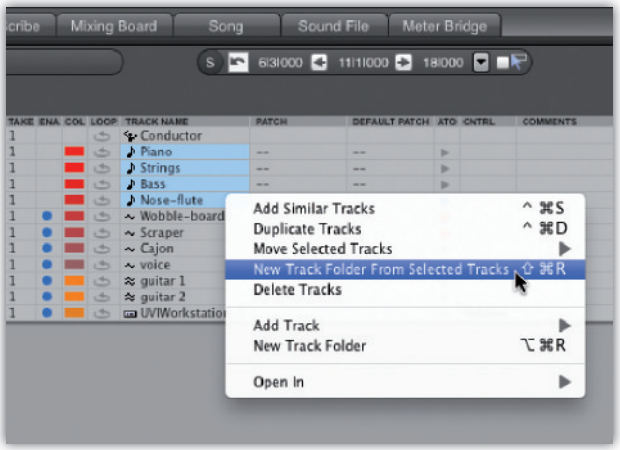
As for the Range Trim modes, these allow you to tweak the values of existing automation up or down within the selected time range. Again, it's a really nice alternative to doing this graphically, with the mouse.

If you really get into this new feature, it's worth noting that you can define the time range in several ways. Making a time-range selection on multiple tracks, as I suggested above, is good, clear way of doing it. But in the absence of any selected data or a time range, DP will use the Memory Cycle region itself. That could save a step or two, depending on how you like to work.

Keep It In Context

For many years, DP has been one of a very small group of applications that didn't use contextual menus, those handy pop-ups that appear when you right-click your mouse, and which offer a streamlined range of commands suited to whatever you happen to be currently doing. Well, no longer, because with version 7.2 contextual menus in DP are a reality. It's a great enhancement — for example, right-click in or below the track list in the Tracks Overview and you get a menu that includes options to Add, Duplicate, Move or Delete tracks, or open individual tracks in different editors. Right-click on a selected soundbite in the Sequence Editor and you can quickly apply any number of Edit and Region menu commands. And that's just the tip of the iceberg.

There are a couple of little things to watch for, though. First, you only need to click near selected MIDI notes, a little above or below them, to bring up their contextual menu. Don't waste time trying to be super-accurate with your mouse and clicking precisely on the events themselves. Second, if you're holding out against the 21st century and still have a single-button mouse, clicking it while holding down the Control key will do the same thing as a right-click. But watch out: there are certain data-manipulation techniques that themselves utilise the Control key, an example being 'throwing' soundbites left or right in the Sequence Editor (which you do by control-dragging them). When there's any potential conflict, the old, long-standing behaviour is prioritised, so in this particular instance, control-clicking will never give you a contextual menu.



Contextual menus, implemented throughout DP 7.2, are a brilliant workflow enhancement — and not before time!

Tiny Tricks

Here's a trio of other great little DP7 enhancements to try on your next project:

- **Scroll Play:** Use your mouse's scroll-wheel (or other scrolling function) to adjust Mixing Board parameters by pointing to them and then scrolling. Just select 'Enable Mouse Wheel for Sliders and Knobs' in the Mixing Board mini-menu.
- **New Seeker:** If you want to check a DP keyboard shortcut, or quickly define a new one for a particular command, use the Command Window's new live search field. Just type a command name there and every associated shortcut is automatically shortlisted.
- **Pitch Perfect:** Using pitch automation on audio that was not tuned to A=440Hz? Now you can reset the reference pitch for middle-A in the Preferences windows' Data Display pane.

Live Beat Repeat

LEN SASSO

We show you the most modest of feats: a wonderful way of repeating your beats.

Getting a handle on Live's Beat Repeat audio effect plug-in can seem daunting at first, but, despite its array of options, the basic concept couldn't be simpler: grab and repeat chunks of audio at regular intervals. This month we're going to get to grips with Beat Repeat, and then use it to do some serious damage.

A one-bar drum loop is shown at the top. Beat Repeat repeats the kick-drum hit played on the first beat at the next six 16th-notes, as shown in the second audio clip. The Beat Repeat default settings (shown at bottom) produce a mix of the two audio clips.

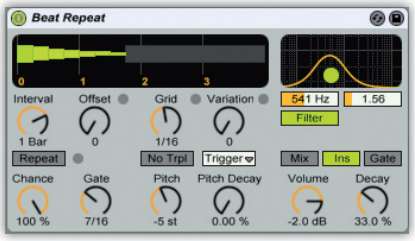
The name 'Beat Repeat' suggests an effect designed for drums, but Beat Repeat is equally adept at enhancing pads and ambient sounds; other rhythm-section parts, such as bass, guitar, and keyboards; background vocals and speech clips; and even leads. I'll use a simple drum loop to describe how Beat Repeat works, then offer alternative examples, along with some tips on how to avoid over-using the effect.

The Simple Truth

To learn your way around Beat Repeat, load up your favourite MIDI drum machine and create a simple one-bar loop with a different sound on each eighth note. Insert a Beat Repeat after the drum machine and play the drum loop. You'll hear the sound that's played on the first beat repeated at the next six 16th-note positions while your original drum sequence plays underneath.

The most important Beat Repeat controls are Interval, Offset, Grid and Gate. Together they determine where (Offset), how often (Interval), and how much audio (Grid) is buffered for repeating (I'll call this the grab). The Grid knob setting is also the repeat rate. The Gate knob sets how long Beat Repeat is active — including the time for the grab as well as all repeats. (If the Gate setting is not bigger than the Grid setting, Beat Repeat won't have any effect.)

In this example, Beat Repeat makes one pass every bar, as indicated by the Interval setting of one Bar (red). On each pass, Beat Repeat waits four 16th-notes then grabs one eighth-note of audio, as indicated by the Offset (green) and Grid (blue) settings. Beat Repeat then repeats the grab with eighth-note spacing, as indicated by the Gate (purple) setting of 6/16: one eighth-note (2/16) for the grab and two eighth-notes (4/16) for the repeats.



The graphic at the top left of the control panel shows the effect of the Volume and Decay settings: the second green bar illustrates a 2dB drop for the first repeat, and the remaining bars illustrate 33 percent volume drops thereafter. Notice the faint grey bar in the graphic; this shows the output level of the source audio and reflects your choice of the Mix, Ins, or Gate button. The Volume, Filter and Pitch settings affect only the processed audio.

Next, notice the Volume and Decay knobs at bottom right of the control panel and the Mix, Ins (Insert), and Gate buttons above them. Volume controls the relative volume of the repeats but does not affect audio passing through the plug-in. Decay determines how much successive repeats drop in volume. The graphic at the top left reflects both settings.

The Pitch and Pitch Decay knobs to the left of the Volume and Decay knobs have a similar impact on pitch. The filter section at the top right processes the repeats with a band-pass filter (again, the source is not affected). You can drag in the graphic or use the numerical controls below to set the filter's frequency and bandwidth.

If you click the Ins button, the source is suppressed when Beat Repeat's gate is open, so you won't hear the original drums during the repeats. If you click the Gate button, you will never hear the source; only Beat Repeat's output gets through. That's the setting you'd most likely want when you use Beat Repeat as a send effect or when you use several Beat Repeats in parallel (more on this later).

Less Is More

Repeating in the same way over and over gets a little boring, and Beat Repeat

In this Audio Effect Rack, four Beat Repeat chains add a 16th-note repeat at or near each beat of a bass loop. Each of the Beat Repeats is in Gate mode, and the top (empty) chain is used for the unprocessed loop. Slight pan offsets add a spatial element.

has three ways to liven things up. Firstly, the Interval knob sets the length of the repeat cycle. Return to the default settings (reloading Beat Repeat using the Hot Swap button is the fastest way), and change Interval to two or four bars. Now you'll hear the repeats of the first beat every other or every fourth bar. Change Interval to 1/2 bar and things get busier—beats one and three now get repeated.

Long Interval settings add breathing space, but the process is still always the same. On the other hand, the Chance knob sets the probability that Beat Repeat will kick in. At 50 percent with a one bar Interval setting, for example, you'll get the effect only half the time, but it won't be the predictable every-other-bar that it is with a 100 percent Chance setting and a two-bar Interval setting. Of course, you can use small intervals and low chance together to produce infrequent, unpredictable repeats.

The Variation knob is your final way around predictability. It varies the grid at specific times, in a manner chosen by the drop-down Trigger menu below it. The knob's value indicates how far either side of the Grid knob's value the grid can vary. The 'No Trpl' button overrides triplet values (1/3, 1/6, and so on) by using the next lowest Grid setting, and that applies to values set with the knob or by variation.

A Different Drummer

To finish off, let's look at some creative ways to use Beat Repeat to process sounds other than drums.

You can add some interest to a repetitive bass line by inserting occasional double hits using multiple Beat Repeats with short gates, different offsets, and reduced Chance settings.

- Create an Audio Effect Rack on the bass track and insert an empty chain to use for the unprocessed audio.
- Drag a Beat Repeat into the rack to create a new chain.
- Set the Beat Repeat's Gate to 2/16, its Grid to 1/16, and its mode to Gate.
- Start the bass loop and adjust the Offset knob until you find a position that produces a useful double hit (zero almost always works).
- Repeat the process with two or three more Beat Repeats, choosing a different Offset setting for each one.

To keep the extra hits from going over the top, you'll want to adjust both their Chance and Volume settings. To that end, map the Chance knobs to the Effect Rack's top macro knobs and the corresponding chain levels to the bottom macro knobs. Now you can dial in the treatment without manually selecting individual chains, and you can easily automate changes; for example, zeroing some chain volumes at different points in your song.

Beat Repeat often comes in handy for adding harmony and reverb to background vocals. To do this, insert the Beat Repeat, followed by your favourite reverb plug-in, into one of Live's Return tracks and temporarily set the send bus to pre-fader so that you can lower the vocal while setting up Beat Repeat. For this effect, you'll generally want 100 percent Chance and no Variation to ensure that the effect is consistent. Grid settings of 1/8 or 1/12 with a Gate setting of 7/16 or 8/16 generally work well. This is a situation in which small Interval settings and consonant pitch shifts often work well, the small Interval setting causing several different positions in the vocal loop to be grabbed, and the pitch shift adding a little harmony.

Pads make good fodder for regular repetition. Set Chance to 100 percent and use small Interval and Grid settings with slight pitch and volume decays. Set Beat Repeat to Ins mode so that the repeats replace the incoming audio; otherwise they're likely to be obscured. A Gate setting below the Interval setting but above the Grid setting lets you hear the unprocessed pad before and after each repeat cycle. For example, with half-note intervals, an eighth-note grid, and a three eighth-note gate, you'll hear the grab (unprocessed pad), followed by two eighth-note repeats followed by another eighth-note of the pad on each cycle.

A pad triggered every two bars (top) is grabbed and repeated with slight pitch and volume decays at half-bar intervals (bottom). The repeats replace the incoming audio (Ins button) and the repeat grid varies between 16th-notes and eighth-notes — one position each side of the Grid knob's 1/12 setting.

As you can see, Beat Repeat will deliver occasional, subtle accents or completely mangle your audio, and both alternatives are useful. Next time one of your tracks goes a little stale, let Beat Repeat mix things up a bit.

In this setup, Beat Repeat grabs the second and fifth eighth-note in each bar of a background vocal, pitch-shifts it down a Perfect Fourth (five semitones), and feeds it into the reverb.

Back to Basics

Whether you're an experienced or a novice Logic user, going back to the beginning might show you a thing or two.

ROGER JACKSON

The sheer scope of Logic can make it daunting for the novice user, and even for the DAW ace there may be quite fundamental aspects of the program that remain unclear. Consequently, we thought we'd take time out this month to recap some of the basics.

If you're a Logic novice, you've bought Logic and fired it up. You've got a good idea of what it does, but now, close up, it's hard to know where to start. Are there some decisions to make and settings to change first? And where do you find everything?

On The Menu

There are two types of menu in Logic: the main menus and the local menus. The main menus are at the top of the screen, and here you'll find items relating to the whole project, such as opening and saving files, window selection and the misleadingly named 'Undo History'. Then, in the Arrange and Editor pages, there are local menus where you will find options relating to the function of each particular window; for example, settings related to tracks and regions on the Arrange page and those related to graphics and music fonts on the Score page — you get the picture! We'll look at some quicker ways to access these in a minute.

Single File!

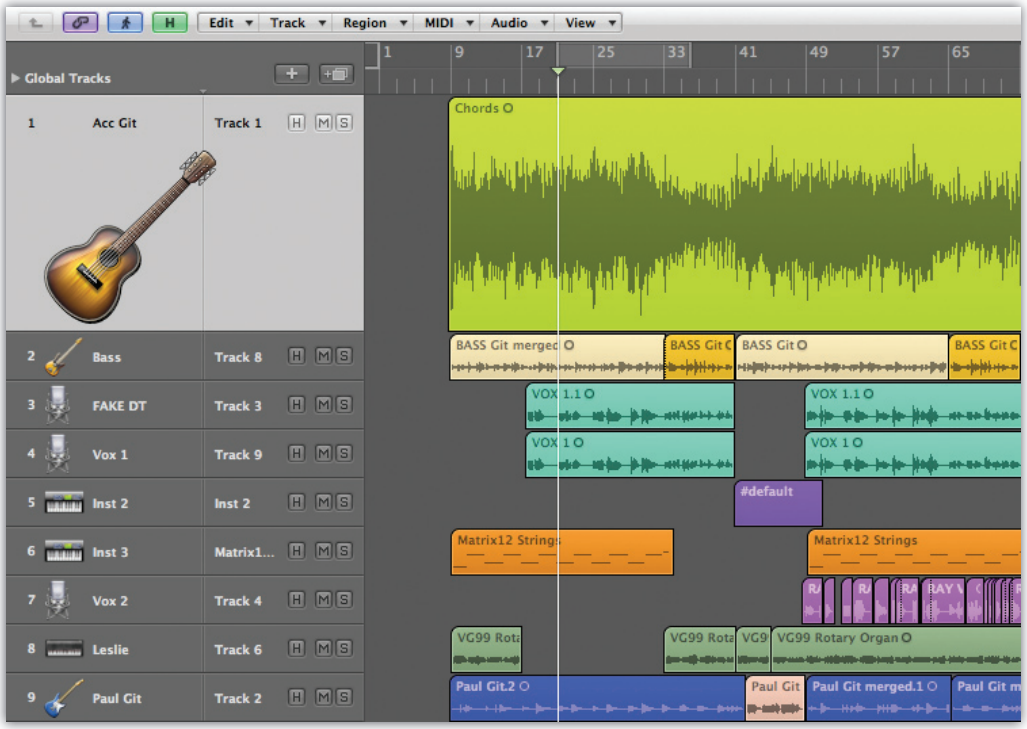
When you start Logic up and select File / New, you'll be offered a variety of templates — basically, songs that are pre-configured for a particular purpose. The ones you get with Logic are designed to demonstrate various features of the application and not generally intended as a starting point for a song. What you need for your new masterpiece is 'Empty Project'. However, you'll probably want to make your own custom templates with a few favourite instruments and a couple of audio tracks ready to record a quick guitar and vocal, for example. (To do this, go to File / Save as Template). Like Microsoft Word '.dot' files, templates are protected from overwriting.

If you've used earlier versions of Logic Pro or Express and have simply carried over your default song, you could be inviting problems, due to differences in the way the mixer now works, so I strongly recommend you create a new template to take advantage of the newest features. Try to store the 'cleanest' template you can, as it can be very frustrating having to weed out sends, inserts and especially automation, which were great on one project, but could make their presence felt in a baffling way as you start mixing.

Logic will ask you where you want to store this song, and it will default to the Logic folder in your home Music folder on your startup disk. This may not be the ideal place, and here's why…

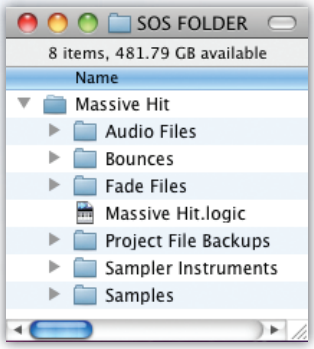
Back in the days when no one had heard of DAWs, an early ancestor of Logic emerged, living entirely on a diet of MIDI. He and those like him were called sequencers, as all they did was put things in sequence. This is essentially what your Logic song still does, although now it has the accumulated baggage of audio files, software instruments and effects settings. Similarly, sampler instruments need their family of samples to go with them wherever they go, so these need to be kept together. We thus end up with a raggle-taggle band of mutually dependent files that make up our Logic project. The whole household is typically made up of the following:

- The Logic song
- Audio files
- Bounces
- Fade files
- Project file backups
- Sampler instruments



In much the same way as any other DAW, Logic uses several different track types(Audio, Software Instrument and External MIDI) and produces 'Regions' alongside them as you record.

Making sure that everything is saved in the correct place is important, as your Logic project requires all manner of files to be saved alongside it. With every song you'll find accompanying audio files, bounces and fade files, for example, that will need to be accessed by the project file.



- Samples
- Undo data

You might also add to that a movie, lyrics, and anything else that the obsessive compulsive inside you tells you to keep in a safe place.

With much of this needing to stream quite rapidly from the disk, it makes sense, if you can, to designate another disk as a work disk, where you are not competing with the computer's own system for disk access. After that, Logic will look after its own filing while you get on with the music.

Tracks & Channels

There are three types of track in Logic:

- **Audio:** For live sounds, whether through a mic or plugged into the interface. This is also what you want for loops.
- **Software Instrument:** This gives you a visible, editable track of MIDI data, while the resulting sound — from Logic's synths, organs, Ultrabeat and EXS24 sampler — is treated in the same way as an audio track.
- **External MIDI:** Although this will address your external synths and racks, it doesn't have any provision for returning their output into Logic's mixer. If you want to use an external keyboard or rack sound source, you should use a software instrument track and, from the list of instruments, select External Instrument. A little dialogue box will ask you for the MIDI number of your instrument and which input channels of your audio interface you have plugged it into. This turns it into a hybrid track sending MIDI and streaming the audio from your instrument into the Logic mixer.

In Logic's mixer you can have five types of channel strip. As well as the aforementioned inputs — Audio, Software Instrument and MIDI channel strips — there are Auxes and Outputs. Auxes are used as buses for sub-grouping things that you might want to treat as one, like backing vocals or a drum kit, or for hosting effects, such as reverb and delay, that you might want to apply to several things in different proportions (for example, having one instance of Space Designer on an aux that you send to from lots of channels is more efficient than having lots of instances of Space Designer on individual channels). Just click on a send and select a bus from the pop-up menu to create a ready-configured aux. In the same way, if you select a new output on any channel strip, that output will magically appear at the right-hand end of the mixer.

It's worth remembering that although the channel-strip meters show input signal levels, the fader controls what you hear out of Logic, not the record level. You have to set that with the knob on your interface (or mixer, if you're using one.) »

You can set up an External MIDI instrument to behave in a similar way to a Software Instrument.

» You will also have seen the inscrutable 'Master' fader. You only need this when you are mixing in surround, so you can just click the 'Master' button at the top of the mixer to get it out of the way until then.

The Inspector

On the left of the Arrange page is the Inspector. Although it takes up valuable space on your screen, it has some handy shortcut features. The two faders at the bottom are for the track and the aux or output that it's routed to, so a lot of the time you don't need to open the mixer. If you click on a send button, the second fader turns into that send's destination, so you can quickly change or adjust the relevant effect without leaving the Arrange page. At the top left-hand corner of the toolbar is the Inspector button to toggle the Inspector should you need to reclaim that extra bit of space.

Once you've created your channel strip, you can select a software instrument from the pop-up list in the I/O panels just above the fader. Or, for an audio channel, set the input and you can modify that 'raw' sound by inserting effects. Alternatively, by clicking on the 'Setting' button on the channel strip, you can avail yourself of ready-made channel strip settings. To get you started, Apple have included many possible combinations of instruments and effects. You can store, recall, copy and paste these and, perhaps more usefully, store your own channel strip settings here.

With tracks installed, you can go for a take. As you record, a block appears on the track. This is a known as a region and can represent audio or MIDI data. You can copy and move these wherever you like, although a MIDI region won't play in an audio track, and vice-versa. You can make an alias of it (Alt-Shift-drag), which will reflect any changes you make to the original, or make it loop (tick the box in the Inspector, or mouse up to the top right-hand corner of the region and, when the cursor changes to a circular arrow, drag to the right). This loop can be any length; it doesn't have to be a multiple of the length of the original.

Tools Of The Trade

The tools are pretty self-explanatory, but the 'Swiss Army Knife' of them all is the Marquee tool. Just as in any graphics application, dragging with the marquee tool defines a free-sized area (covering multiple regions, if you like) for further operations. You can now do a variety of things with the selected area: set locators, play just this area, punch-in record, mute, cut and copy, along with a thousand (or so) more operations.

If zooming is something you like to do (and I think you do), note that if you drag out an area ('rubber-banding' it) while holding Control-Alt, that area is enlarged to fill the screen. Holding Control-Alt and clicking zooms you back out to where you started.

Help Yourself!

A lot of new user enquiries are what those in the support business call 'RTFM' issues, whose chief characteristic is that the solution can be easily be found by consulting the included documentation. But since Apple have commendably adopted a greener packaging strategy, Logic no longer ships with a bulky tome of instructions.

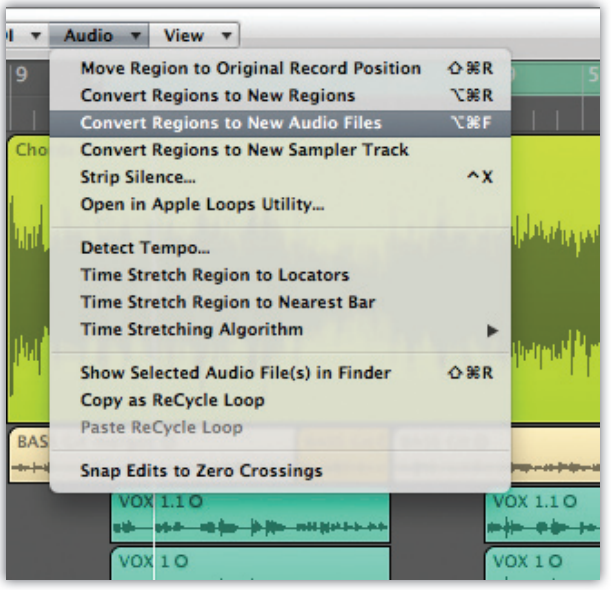
You do still get one, though. It's there under the Help menu. Although this won't help you if what you need to learn is how to launch Logic itself, for other enquiries it is invaluable and offers a fast-paced introduction, as well as dealing with the instruments and effects in detail. At the risk of appearing pedantic (moi?), I suggest that a few hours exploring here will save weeks of your time later. It's great, too, to have this searchable document at hand for when problems do arise. The Help menu also directs you to Apple's discussions on the web, where you can share your frustrations (and solutions) with others. It's well worth making friends with the very knowledgeable and generous community at the Logic Users' Group (www.logic-users-group.com).

Save & Prosper

You can set Logic to keep the previously saved copies of your song — up to 100 of them (look in Preferences / General) — just in case you discover some earlier blunder and need to scuttle back. This is very reassuring, as is the Undo History, which goes back to when you launched the song. You can't undo selectively, however, only return to a previous snapshot state, which might also involve undoing some actions that you would otherwise like to keep. Some audio functions are not undoable, though, and since you're actually altering the original audio file, it can be worth your while creating a copy to work on as you perform a destructive function. To do this:

1. Cut out an area from the audio region you want to change.
2. Select Audio / Convert Regions to New Audio Files.
3. If it doesn't sound right, you can use the Undo History to go back to that blissful state before you began the whole sorry process.

I have found it useful to develop a nervous left hand that hits Command-S whenever I make any changes — every five or 10 seconds, I should say — as trying to retrace »



For extra backup you can convert specific regions to audio files. This is a good idea when you're about to perform a destructive function, as it gives the option of restoring exactly what you were working on, rather than using the Undo function.

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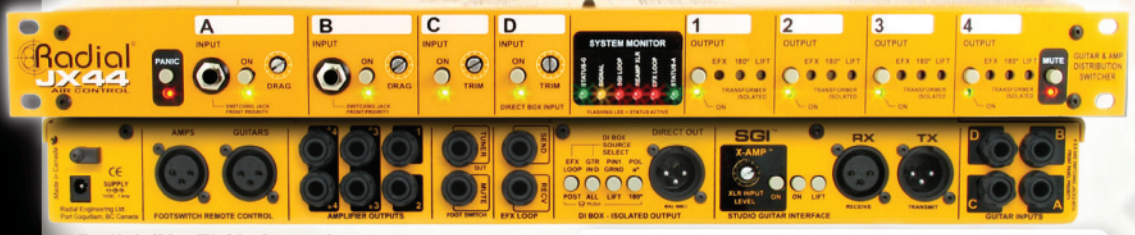

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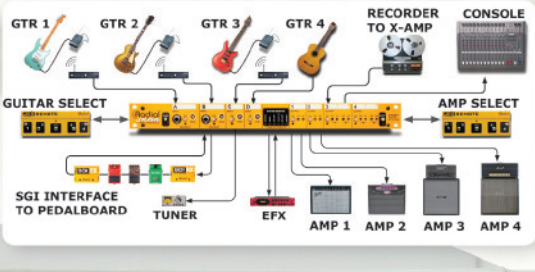

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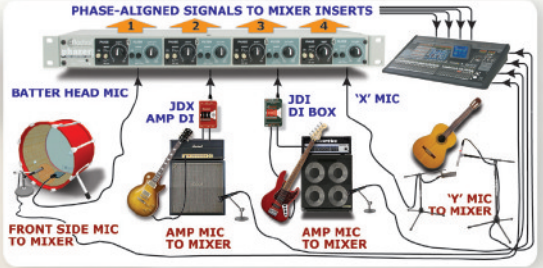

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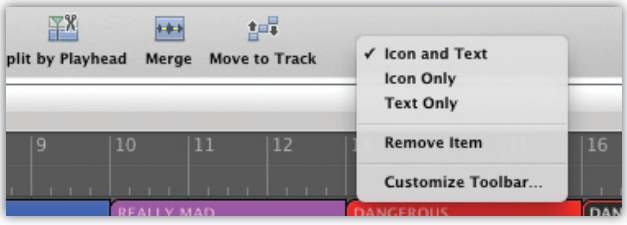
» your steps since the last save, following a crash or freeze, always leaves you feeling that it's not as good as you had it before.

Settling In

The extent to which you can customise Logic can be quite daunting, but that shouldn't prevent you from trying a few adjustments to your working environment. Key commands and screensets can be a good starting point.

It can be very slow and long-winded to use the mouse and menus for every operation, and while you will have quickly found the key commands for the transport controls, there are plenty of reasons to explore the Aladdin's cave that is the key commands page (Alt-K). You'll find commands for things that you didn't know Logic could do, even for things that you can't imagine a need for. But you'll also find some things that delight you.



Logic's user interface is highly customisable. From small actions, such as is shown here, to completely changing what appears in the toolbar, you can really make yourself at home!

Note that you can select commands by 'key label' or 'key position'. This means that you could have, for example, numbers 1-0 at the top of the keyboard to take you to Markers, while the numbers on the number pad perform a different function: recalling screensets, for example. I also have a key for dropping markers on the fly at important song points, giving me a snappy way to navigate the song. If you want to move a marker, or drag it out of the timeline entirely, you will need to hold down Alt first.

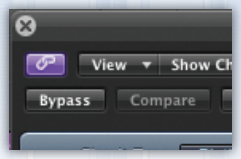
One of my favourite key commands is 'Play From Selection', but your method of working will determine exactly which commands you find most useful. The Toolbar at the top of the Arrange page is a place to keep buttons of your favourite functions to hand. Click and hold in the Toolbar area, and select Customise Toolbar from the menu that falls open under your enquiring touch.

You probably know that screensets enable you to recall the size of various pages in a layout that you find useful. If you're working with limited screen space (and who isn't?), you can easily flip between different views, or instantly tidy up your workspace after you've been editing or browsing lists and libraries.

But that's not all: at the top right-hand corner of the Arrange and Editor pages lie two tool icons. These are the default left-click and Command-click tools and, at Preferences / Editing, you can choose to add a third for right-click. There is a different selection for each editor too. Choose the ones you use most in each page and then save the screenset so that they come back next time you need them.

There is an Auto Zoom button in the toolbar that zooms just your selected track, saving both space and eyestrain. Click and hold the bottom left of the track header (where the open hand changes to a pointing hand) to adjust the amount of zoom.

Another clutter buster is to limit to just one the number of effects windows open at any one time. Open an effects window and click on the link icon in the top left-hand corner — it will glow purple — and as you select another effect, the window of the first one will dutifully vanish.



Clicking the link icon in the top of an effects window will mean that, when you open a second effects window, the first will close, thereby saving valuable space on your screen.

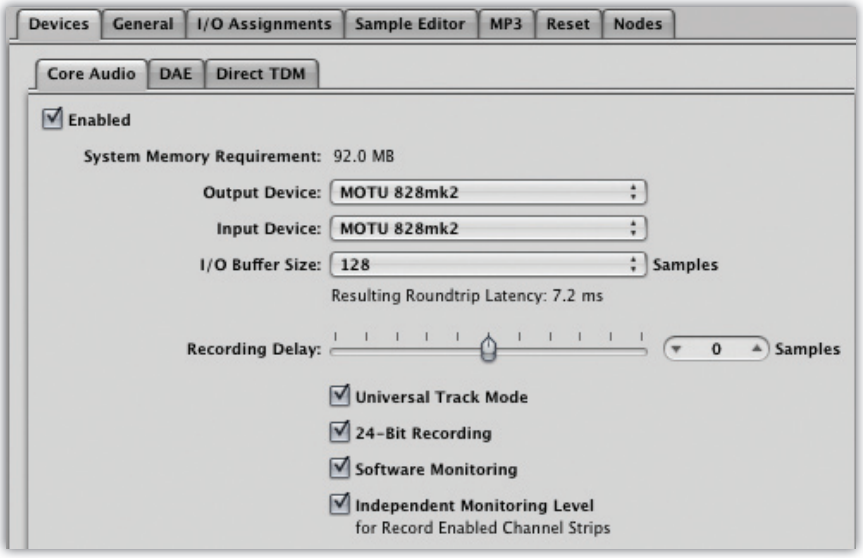
Ready, Set...

As for setting up your Logic system, you will, for the most part, be quite alright getting started with the default values. However, one that you might want to check is the I/O buffer size. This is a bit like the amount of cash that banks are required to hold against the risk of liquidation, in that if the buffer is too low, you risk instability problems! Playback also might stutter, or you could start to hear clicks and pops. If the buffer is too high, you get problems with latency (the delay you experience when recording, between the live signal and what you hear back once it's passed through Logic). Ideally you would have the buffer lower when recording and higher when mixing. Realistically, though, you want to just set it and forget it. Go to Logic (Pro or Express) / Preferences / Audio / Devices / Core Audio and try a setting of 128. You can move up to 256 if you start to notice pops and clicks.

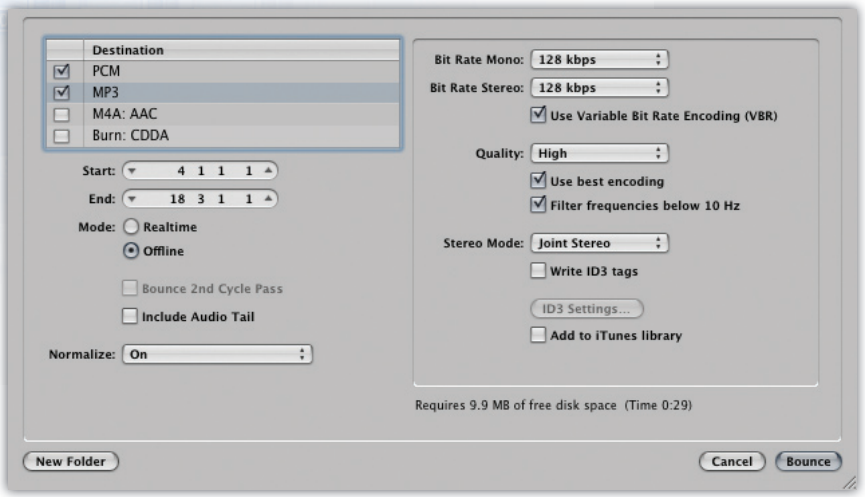
A few other settings may catch your eye on this page, beneath the Buffer Size drop-down menu:

- **Universal Mode:** Using this means that you can have stereo channels instead of two linked monos.
- **24-bit recording**: Why wouldn't you?
- **Software monitoring:** Leave this ticked unless latency is driving you mad, in which case you will have to rig up an alternative way of monitoring your live input with the already recorded track.

Settings in Preferences apply to Logic universally, but similar controls may be



If you're suffering latency problems, you might try lowering your buffer settings. Ideally,the buffer would be lower for recording and higher for mixing, but setting it at 128 is a good start and you might be able to just leave it there. If you start to hear clicks and pops, try moving it up to 256.



When bouncing your tracks, tick MP3 in the Destination list to create an MP3 file alongside any other format you're creating.

found in File / Project settings, which are saved with and apply only to your current project. These include whether to save used Sampler Instruments and their samples with the project (which you only need to do if you're going to work on it elsewhere), the amount of count-in before record, and the sample rate for recording. For most of us in the UK, 44.1kHz is fine, but you can up this as far as 192kHz if you have disk space to spare and the marginal quality difference is important to you.

Note that this is only for Logic's internal use, and when you bounce to create your master, other considerations apply:

- **Output format:** What you output depends on what you or your client are going to do with the mix. If it's going to CD, 16-bit WAV or AIFF at 44.1kHz is what you need; otherwise go for 24-bit. TV broadcasters like 48kHz, and WAVs have a time stamp so can be easily laid into a soundtrack edit. Also, choose interleaved unless you're sending the mix to a Pro Tools user.
- **MP3:** If you want an MP3 to send to a friend, Logic will bosh you one out at the same time if you check the box. A setting of 128 to 160 kbps is a reasonable trade-off between compact file size and unlistenability.

Logic has grown into a hugely versatile and immensely configurable application that seems to reflect each individual user's personality. We all have to discover our own way, but if you have previously encountered mists of confusion, I hope you can now discern a track that might bring you nearer your own personal Nirvana.

Safe & Sound

In the first of this two-part series, we look at the best ways to manage your day-to-day backups in Sonar.

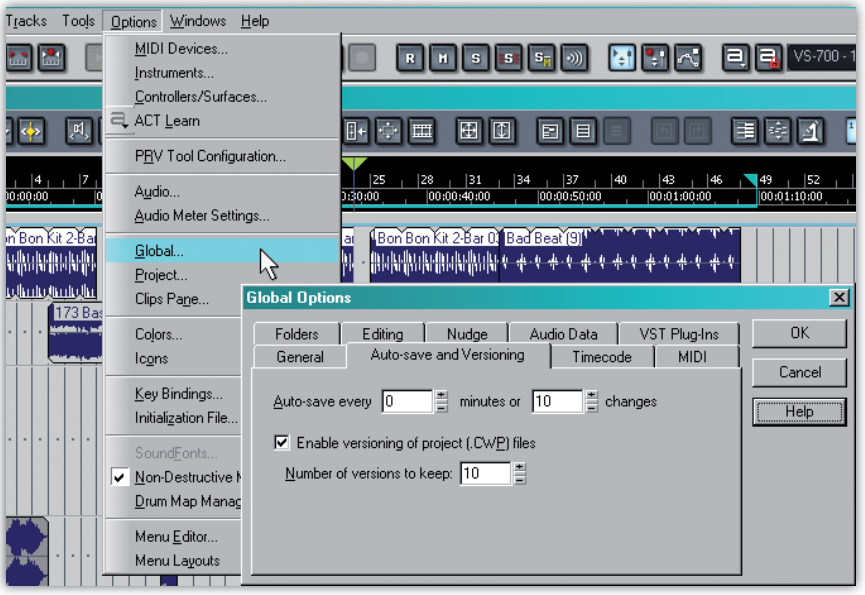
CRAIG ANDERTON

The most important item in your computer isn't the CPU or motherboard, but the data. And if there's one thing I've learned, it's that you never know when some project you did years ago will need to be resurrected. If you consider yourself even remotely professional, you not only need to save your work, but also make sure that you'll be able to use it at some time in the far-away future.

In this two-part series, we'll first cover the conventional way to do backups in Sonar, and in the next installment we'll branch out into the next level: archiving projects. So that we're clear on definitions, we'll consider backups to be saving your work from day to day, while archiving is 'putting something on the shelf' when you're done, which requires longevity as well as security.

Filing Better

First you need to know what you're backing up. The main Sonar .CWP project file contains no audio, as it's a 'roadmap' for the project — for example, paths to audio clips, where clips sit on the timeline, MIDI data, which plug-ins are used, and so on. So, while saving this file is vital, you also need to save the audio to which it points. Also, samples used by virtual instruments aren't saved as part of the file. As long as the instrument and effects plug-ins are installed on your computer and Sonar knows where to find them, loading the .CWP file will load the appropriate plug-ins and sounds. If Sonar can't find these, you're in trouble, but we'll cover possible preventative measures in part two.



Auto-saving is very convenient if you're the kind of person who gets so involved in your work that you forget to save periodically.

Sonar has three main ways to deal with the audio-file part of the puzzle:

The Global Audio Folder. When hard disks were expensive and had limited capacity, Cakewalk introduced the concept of a Global Audio Folder in which all audio for .CWP project files lived. Then you could just back up that folder and know all your audio files were safe. However, with the low cost of hard drives, optical media and even USB sticks, the Global Audio Folder is pretty much obsolete.

The Bundle File. When you 'Save As' a project file, under the 'Save as Type' drop-down menu you have the option of choosing the Cakewalk Bundle (.CWB) format. This single file incorporates everything stored in a normal project file, as well as all audio (but not video) used in the project. While convenient and portable — and a great way to transfer works-in-progress to collaborators — if the bundle file becomes corrupted, you lose the entire project.

Per-project Audio Folders. With Per-project Audio Folders, you can save all audio, along with the project file using that audio, in its own folder. Back up the folder and you back up the entire project (with the exception of plug-in sample-library data, as mentioned earlier). Unlike a bundle file, if one of the per-project files becomes corrupted you can still use the remaining ones. I use and recommend the per-project approach, so let's investigate it further.

Per-fect

We'll begin with some prep work. Go to Options / Global and select the Audio Data tab. Tick 'Always Copy Imported Audio Files'. This is useful if, for example, you load a loop from a CD-ROM: that loop is copied and made part of the project audio, so that even if the CD-ROM isn't in the drive, the audio is still available. Also tick 'Use Per-project Audio Folders', as that automatically sets up the file structure we want: a folder for the project containing the .CWP project file and an Audio sub-folder that holds the audio used in the project.

Now when you Save, all audio goes into the audio folder, and when you call up the .CWP file, it knows to look in that folder for its audio. Note, however, that we still haven't taken care of what happens if a virtual instrument or plug-in used in this project is uninstalled prior to calling up the file at some point in the future.

We'll describe how to deal with this in the next issue.

You can easily verify that the project file is working properly and storing the right information. Go to File / Project Audio Files to open a window that displays the path for the project folder, the project file, and the audio folder. It also shows all files that are used in the project, along with their file path, bit depth, file size and status. All status entries should read 'Local', because any audio from an external source should have been copied to the hard drive. If an entry doesn't show 'Local', you need to make sure that 'Always Copy Imported Audio Files' is ticked.

External MIDI Data

If you're using external MIDI gear — instruments, controllers, effects or whatever — you should save any presets from that gear as well. Fortunately, you can save SysEx data within a Sonar .CWP project file. This is extremely convenient, as you can then consolidate your MIDI data with the rest of the project. To do this:

- Go to Options / Global / MIDI and make sure System Exclusive is ticked.
- Now go to View / Sysx, which brings up a window that shows 8192 banks of SysEx. A bank could be anything from one program to a synth's complete internal memory.
- When you click on the window toolbar's red arrow, a window appears with a list of supported MIDI gear. If it shows the device you want to back up, select it and click on OK. This initiates the SysEx recording procedure.
- If the device isn't listed, select 'You start dump on instrument' and click on OK.

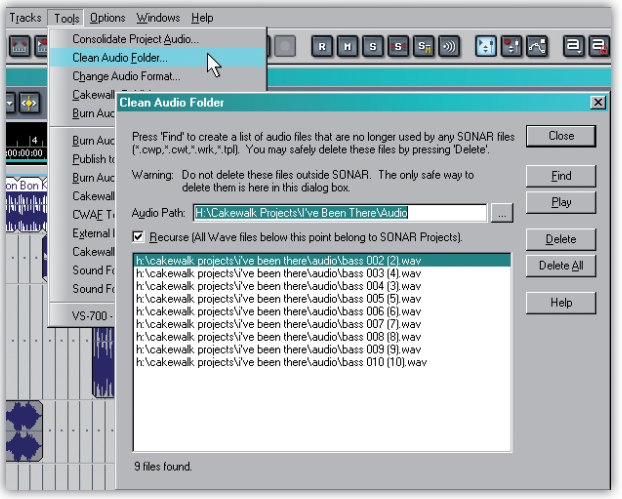
Initiating a dump also initiates SysEx recording. A counter increments as Sonar receives the data. When it stops, the process is done. This data is saved with the project and available when you load it. If you ever need to recreate the MIDI setup you used for a project, click on the toolbar's black arrow to send the SysEx to the MIDI device. (You can also save the file to disk, as well as set up automatic sending of the bank when you load the project. Check out Sonar's help files for more info.)

Spring Clean

Before archiving a project, it's worth deleting any unused files, not just to save space, but also to avoid confusion at some later date ("What's this file called "Skeezix" doing here?"). It's easy to build up a bunch of unused files if, for example, you do multiple vocal takes but then decide to get rid of all of them except the 'keepers'. This could also happen if you import lots of loops into a project to compare them with loops that are already loaded, and then delete most of the candidate loops. Because you imported the files, Sonar will dutifully copy them to the project's audio folder, where they'll stay, unless you use the Clean Audio Folder function. This scans the audio file folder of your choice and presents a list of unused files you can delete.

To use this, go to Tools / Clean Audio Folder. A dialogue box opens up that defaults to selecting the Audio folder for the current project, which is probably what you want. If not, you can browse for the folder. Tick Recurse to include sub-folders within the project folder (although I can't envision a common scenario where you'd have these kinds of sub-folders).

Depending on how much local storage you have, compiling the list of unused files can take a long time, because Sonar checks to make sure these files aren't used by any other projects. Once the list is compiled, you can delete individual files, or all of them.



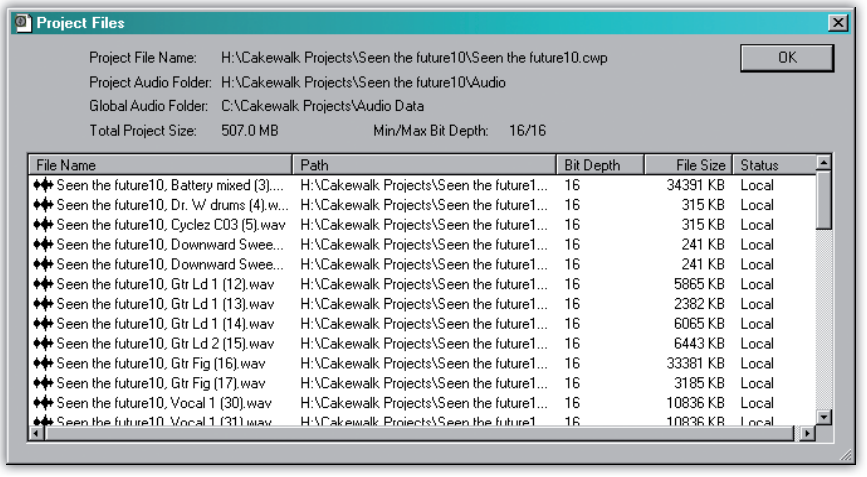
In this project, I had brought in 10 bass loops to hear how they worked with the drums, but kept only the first one. Running the Clean Audio Folder utility located the other nine loops in the Audio folder so I could delete them.

Auto-saving

Sonar can automatically back up .CWP files throughout the course of a project. You access this by going to Options / Global, then clicking on the Auto-Save and Versioning tab.

You can specify auto-saving after a certain number of minutes have elapsed or after a certain number of changes have been made, but it's a good idea to do versioning — where Sonar saves successive versions of the project — as well. This is crucial so that, if you make a change you didn't want to make and Sonar saves that version, you can revert to a previous version. As .CWP files aren't very large (typically a few megabytes for medium-sized projects), feel free to pile on the versions.

We're off to a good start on backing up, but we're not finished yet. See you next issue!



This window lets you confirm the contents of the Project Audio Files folder.

Equal Opportunities

MIKE SENIOR

Discover the power of Reaper's EQ plug-in and become an EQ master in the process.

I use Reaper's ReaEQ plug-in a great deal for general mixing purposes and I think a lot of users underestimate it, so in this month's column I want to explain what I like about it, and in the process pass on some general-purpose EQ tips.

The first thing to say is that ReaEQ will give you as many EQ bands as you want: just click on the Add Band button and then select whichever filter type you like from the new band's Type drop-down menu. This means you can use a single plug-in instance to deal with tracks that require both detailed corrective EQ and more broad-brush tonal shaping. That said, do remember that it's usually a good idea to keep the number of active bands to a minimum when using any EQ plug-in, because that will tend to reduce detrimental side-effects of the processing.

Gain Some Control

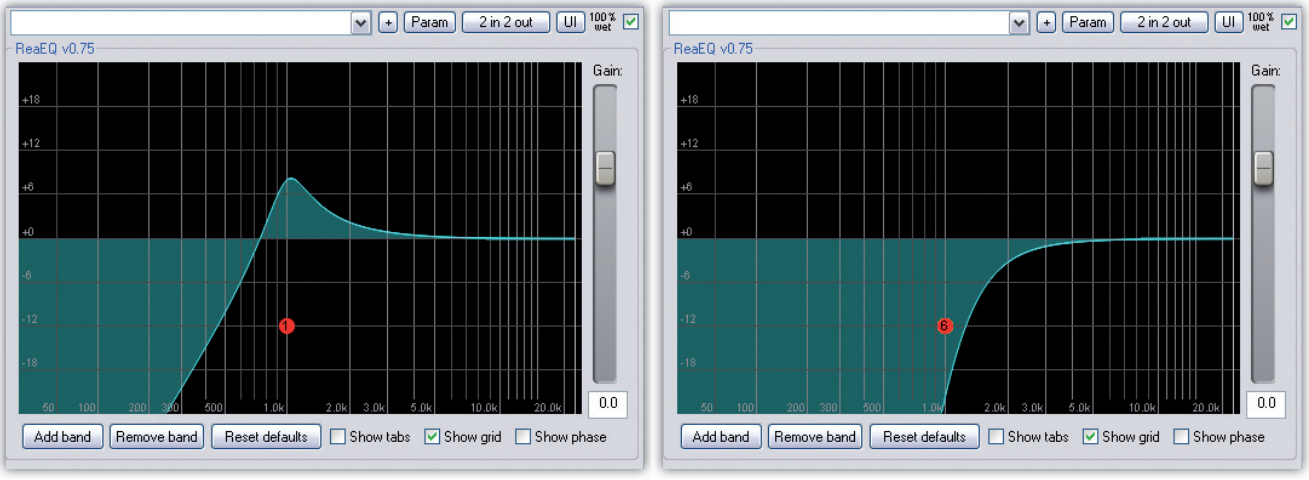
It's also nice that there's an overall Gain control (something a surprising number of EQ plug-ins still don't provide), because that allows you to compensate for any subjective loudness increase you've created through your EQ setting. If any plug-in makes your track appear louder, that can fool you psychologically into thinking that the sound is 'better', irrespective of whether it actually is. Careful use of ReaEQ's overall Gain control can head off this problem, so that you can use the effects bypass button to assess the effectiveness of your settings without playing perception tricks on yourself.

Take The High Road

As far as I'm concerned, any EQ plug-in without a high-pass filter is about as useful as a pint of beer without the glass. This simple type of EQ usually accounts for about half the processing bands on any mix I do, and I usually advise *SOS* readers to high-pass filter almost every instrument as a matter of course. The basic principle is to listen to each recorded track within the context of the whole mix, and then filter out any low end that isn't actually heard in practice. In some cases, such filtering may make some instruments sound painfully thin on their own, but that doesn't matter as long as they sound right when everything else is playing too.

The main reason why so many Mix Rescue submissions suffer from muddiness and lack of clarity is because few home recordists apply systematic filtering. Another reason to scotch any unwanted low end, however, is that low frequencies are what eat up most of your overall mix headroom, so any rubbish in that spectral region can drastically hinder a mastering processor's ability to increase the subjective loudness of your production (something that's a concern if you're wanting to compete in this regard with heavily maximised commercial records).

One of the things I most like about the filter in ReaEQ is that you get a great deal of control over its slope. The first way to tweak it is via the Bandwidth control. Although the



Reducing the bandwidth of ReaEQ's high-pass filter won't make its slope any steeper, it'll just add emphasis and resonance around the cut-off frequency, as shown in the first screen (which uses a Bandwidth value of 0.5). Stacking several high-pass filters, on the other hand, works a treat, as in the second screen, where six bands are combined to create an extremely steep 72dB/octave roll off.

What's Not To Like?

Although I do use ReaEQ an awful lot, there are inevitably occasions when it's not the right tool for the job. The most common situation is where I'm looking for subjective tonal changes, rather than just adjustments of the frequency balance. A more obviously characterful EQ plug-in (often an analogue emulation) usually works better there, or else I forsake the EQ in favour of some kind of distortion or modulation treatment. The other situation in which ReaEQ doesn't work for me is when I need linear-phase EQ to avoid phase-cancellation problems, perhaps when sculpting a multi-miked ensemble recording or a parallel compressor's return channel. Something like DDMF's LP10 is a good affordable alternative in that case.

The Gain slider at the right-hand side of the ReaEQ plug-in interface helps compensate for any loudness increase, which makes your EQ judgements more objective.

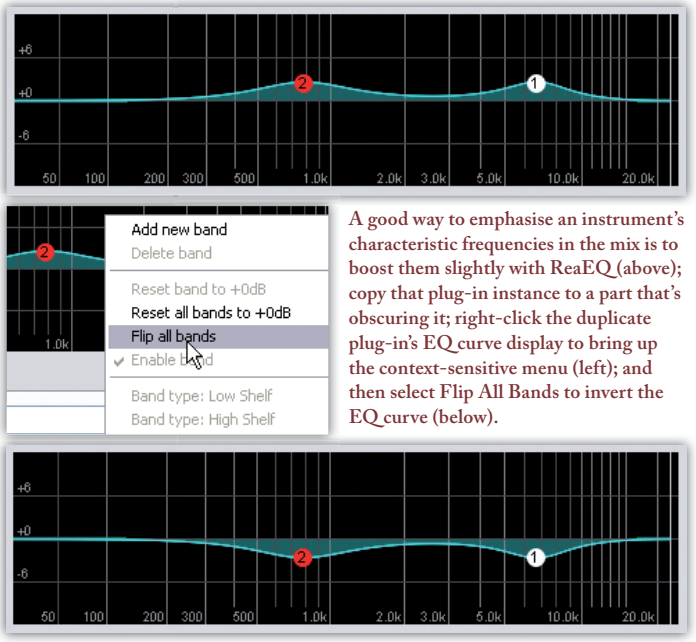
default value of 2.0 is a good general-purpose setting, I do find myself regularly increasing this, effectively softening the filter curve's 'knee'. Why is this useful? Well, for one thing it means that I can use the filter not just for LF rubbish removal, but also as a substitute for any necessary low shelving cut, and thereby kill two birds with one stone. Why take two EQ bands into the shower, when you can just filter and go? In addition, however, there are good reasons to soften the filter knee for wide-ranging melodic instruments, in particular, because it doesn't alter the level relationships between neighbouring harmonics as much, so the instrument's tone won't vary so much as it changes pitch. It's not at all uncommon for me to end up at the maximum 4.0 Bandwidth setting in this context.

By contrast, I also regularly want to steepen the filter's slope, for example when eliminating bass/kick sounds from sampled rhythm loops, or when removing the unpredictable fundamental of a bass instrument so that I can replace it with a sub-bass synth part. Reducing the filter's Bandwidth setting doesn't really do the job here, though, and will just add emphasis and resonant ringing at the filter's cut-off frequency. This is fine for synth patches, but a recipe for low-frequency mess in a mixing context. A better way to increase the filter slope is simply to stack several high-pass filters on top of each other at the same frequency. A single filter will give you a slope of 12dB/octave and each subsequent one will add 12dB to that slope, so a stack of four filters should give you a slope of 48dB/octave, for example.

Peak Performance

After high-pass filtering, the second most useful corrective EQ, in my view, is the narrow peaking-filter cut, which is most commonly used for pulling down unwanted recorded resonances: poorly tuned or damped drums may have pitched resonances that clash undesirably with a song's harmonies, any amplified instrument can suffer from strong speaker-cabinet resonances, and every microphone has the potential to pick up unnatural sounding resonances from poorly treated studio room modes.

One way to find such resonances in a track is to sweep a low bandwidth peaking boost around the spectrum, hunting for frequencies that sound ugly. However, I now usually find it quicker to winkle them out by looking for their tell-tale energy peaks on a high-resolution spectrum analyser. (The ReaFIR plug-in has an analyser that can work well for this, but you may want to increase its FFT Size setting to get a suitably high display resolution. Alternatively, Schwa's affordable Schope plug-in provides a more elegant solution that won't snack as greedily on your CPU cycles.)



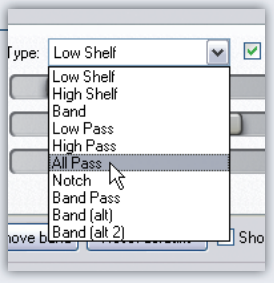
A good way to emphasise an instrument's characteristic frequencies in the mix is to boost them slightly with ReaEQ (above); copy that plug-in instance to a part that's obscuring it; right-click the duplicate plug-in's EQ curve display to bring up the context-sensitive menu (left); and then select Flip All Bands to invert the EQ curve (below).

There are several aspects of ReaEQ that are especially handy in this application. Firstly, its peaking filter can be made extremely narrow, and I'll often use settings of less than 0.1 if I want to target individual sine-wave components, such as those of a pitched drum resonance. The ReaEQ peaking filter will also automatically convert into a notch filter (which completely removes the signal at its centre frequency, rather than just attenuating it) when you dial in more than 24dB of cut. This is useful when a recording has a pitched element to it that serves no musical purpose, as in the case of recorded mains hum, for instance. The ability to create as many bands as you want is also handy here, because it's not uncommon for a badly recorded snare-drum part to have half a dozen unwanted resonances that need in-depth surgery before you even think of trying to address any overall tonal changes.

There are numerous other ReaEQ functions that appeal to me beyond these, though. For example, the way you can use the Bandwidth control to adjust the shelving filter slopes, or the way you can flip the gain settings on all bands (via the EQ curve display's right-click menu). The latter is good for situations where you're wanting to clear space for an instrument's characteristic frequencies, not only by boosting them on its own track, but also by cutting them on conflicting tracks. So even though I now have a lot of great third-party EQ plug-ins on my system, I still use ReaEQ to death.

All Aboard

ReaEQ also incorporates an all-pass filter, a filter type that can appear a bit nonsensical, because it often doesn't seem to affect the tone of the processed audio at all. However, the point of an all-pass filter is that it alters the signal's phase response independently of its frequency response. (If you're not clear what 'phase' is, check out my 'Phase Demystified' article in *SOS* April 2009 for a full explanation. Here's a link: www.soundonsound.com/sos/apr08/articles/phasedemystified.htm.) As such, it's mainly useful where you want to adjust the phase relationships between several tracks, for example when working with multi-miked drum kits, or combining a single instrument's DI and mic signals. Sweeping an all-pass filter on a channel in one of these scenarios will change the nature of any phase-cancellation between the tracks, and thereby deliver a (frequently dramatic) tonal change.



Block Party

Make light work of song arranging with Reason's new Blocks feature.

ROBIN BIGWOOD

One of Reason 5 and Record 1.5's major new features is 'Blocks'. It's a system that allows you to easily construct and edit song structures and repeated accompaniment patterns, making Reason and Record feel less 'linear' and more pattern-based. For those working in musical styles that rely on a lot of repetition, it could be hugely beneficial and a serious time-saver. There's nothing quite like it in any other DAW, though, so read on to find out how it works, and whether it might suit your way of working.

A Record song being constructed using Blocks. The song sections in the Blocks track were first programmed individually. Now, in Song View, they're easily added, re-ordered, duplicated and replaced, to give huge flexibility in experimenting with song structure. Some of the grey block clips, which aren't directly editable here, have been muted to provide localised variations in the arrangement, and various 'linear' material (such as the vocal track) has been added over the top.

A Little Background

First, a brief history lesson. Back in the not-so-distant past, when tape ruled in the studio and computer printers could give you hearing damage, it was the drum machine that formed the backbone of many pop songs. You'd start by programming multiple short patterns, then use an on-board song sequencer to tie those together into larger structures. Entire songs could be created with just a handful of patterns — main grooves, alternatives, fills and so on — replayed in an appropriate order.

These days, the pattern/song approach has mostly become confined to hardware groove boxes (the modern-day descendant of the drum machine) and the dusty corners of the odd DAW. Whatever its potential benefits, it's also got the potential to be somewhat limiting, and, in any case, mouse-driven graphical interfaces have established alternative ways of working.

Now, back to Reason and Record. Never ones to turn down the opportunity to incorporate retro concepts into cutting-edge software design, Propellerhead have resurrected the pattern/song idea in the form of Blocks. Actually, we already had it, on a small scale, with Reason's pattern-based Redrum and the Matrix Sequencer. But Blocks is a much bigger deal: a single pattern (or rather 'Block') can be an entire multi-track song section, such as a verse or chorus. String some Blocks together and you've got yourself an entire song. Want to try a different structure? Just re-order or change the length of the Blocks. Want to use Blocks for a sequenced backing but add linear-style guitar and vocal tracks over the top? No problem. This is a powerful new feature that could change the way you work in Reason and Record for ever more. But it's also very flexible; you needn't use it at all, you can use it a bit, or you can commit to it 100 percent and never work in a linear way again.

Building Blocks

Blocks is simplest to use and understand when you start a new Song with it, and in the new versions of Reason and Record it's enabled by default. You can't really miss the on/off button: it's a great big square in the Transport. With the feature enabled (in which case the button turns blue), a subtle but crucial change occurs in the Sequencer: at top left, two new buttons marked 'Song' and 'Block' appear, along with a new Block track. »

New buttons appear in the sequencer when Blocks is enabled in the transport section. 'Song' and 'Block' switch between Song View, where you lay out your song structures and work with linear material as usual, and Block View, where you work on individual blocks.

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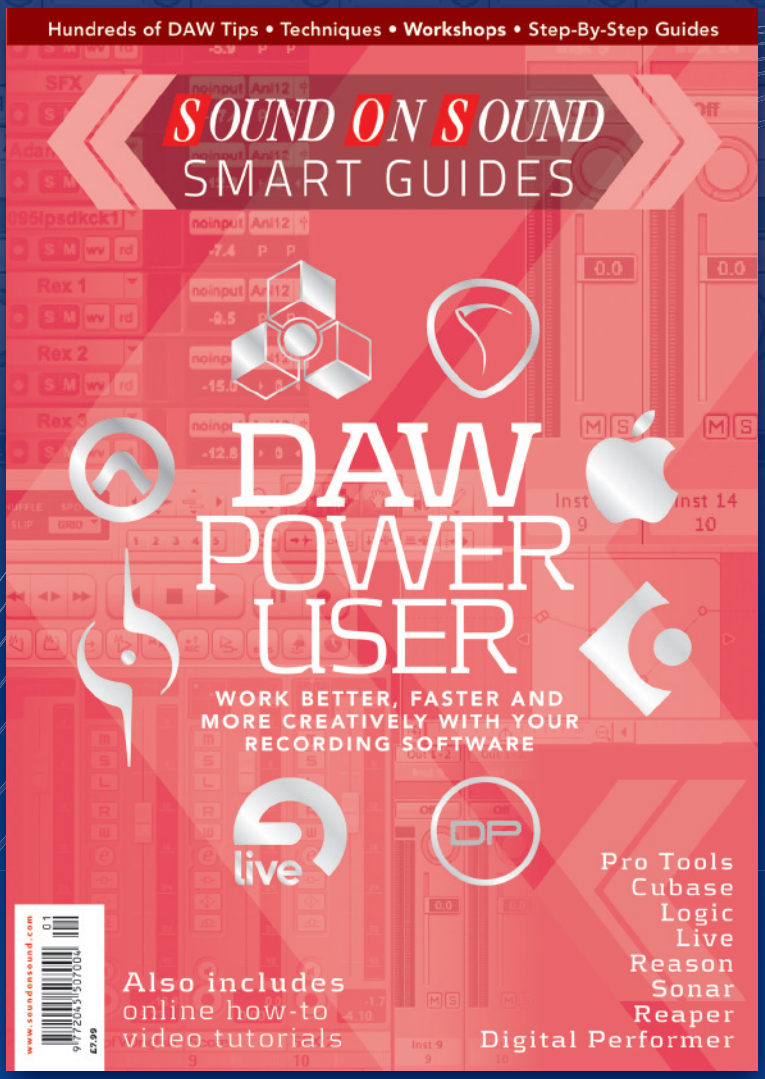

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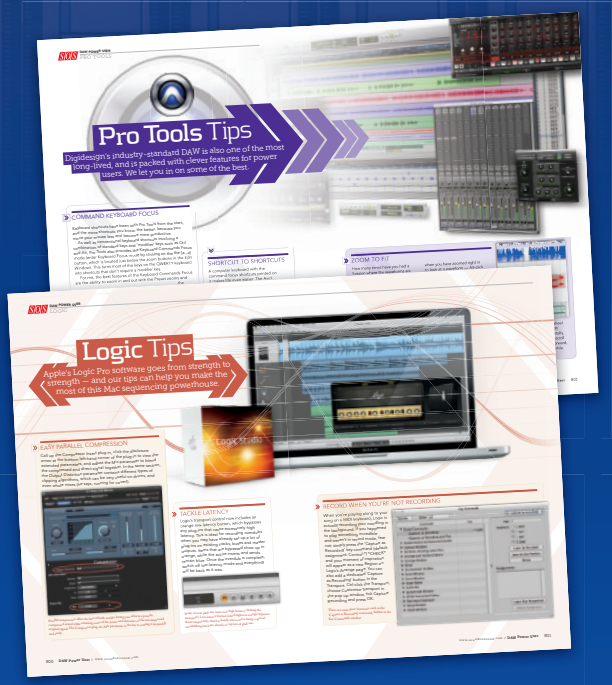

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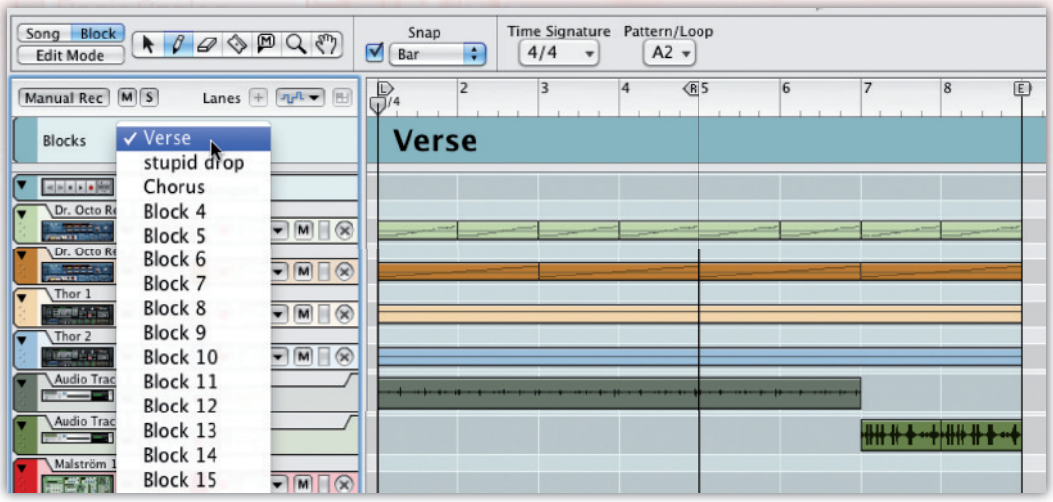


Block View is where you work on individual song sections. The End Marker in the time ruler is crucial for setting the block length. And you choose the block you want to work on from the Blocks track's pop-up menu, as shown.

» Here's how it works: If you click that new Block button in the sequencer, you effectively go into a sort of 'pattern edit' mode: Block View. In many ways, the sequencer behaves the same as it ever did; you add devices and tracks, record and play back as normal. But look at the Blocks track. It's filled with a solid coloured bar that displays a name for the current Block (always 'Block 1' by default for the first one). Also, the End Marker (the 'E' flag in the time ruler) becomes very important, defining the length of your Block and causing the transport to loop back to the beginning whenever it's reached.

So let's say I commit to making a song using Blocks. What's my workflow? Well, to begin with, I might work on an eight bar Chorus section. I'd drag the End marker to the beginning of bar nine to indicate an eight-bar Block length. Then I'd build up my arrangement in the normal way, adding tracks and switching between Arrange and Edit views as necessary. Oh, and at some point I'd double-click the 'Block 1' name in the Blocks track and rename it to 'Chorus'.

Next, I need a verse section. From the Blocks track pop-up menu, I'd choose the next Block in the list (Block 2). The track lanes empty, although the actual tracks remain, and once more I can set the Block length, build my arrangement, and rename this Block to 'Verse'. And so on, then, for any other main sections of my song, such as middle eights and bridges. I might well copy and paste some backing material such as drum loops between Blocks as I go — and that's perfectly permissible.

Song View

What I've got now are a bunch of separate sections, but no sense of running order. This is what Song View is for.

If I click the Song button at the top left of the sequencer, the track lanes empty once more, and that includes the Blocks track, indicating that I'm not working on any specific Block any more. Now comes the fun bit.

After switching to the Pencil tool (click its button, or press the 'W' key), some pop-up menus appear in the inspector section above the time ruler. These include one labelled 'Block'. The idea is that you now draw song sections into the song. So let's say I want to start with two verses back to back. I'd choose 'Verse' from the pop-up menu, and then in the Blocks track I'd click and drag over 16 complete bars. As I do this, the verse section I've been working on begins to fill the individual sequencer tracks, as if by magic. (It's a good idea, by the way, to have Snap enabled and set to Bar before doing this, so you don't have to worry about being too accurate with the mouse.)

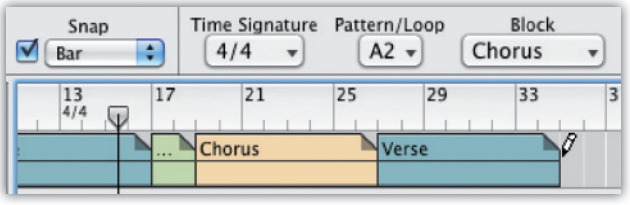
Next, I might choose 'Chorus' from the inspector panel pop-up menu, and draw in a Chorus directly following my verses. And on it goes for the other sections, with a song literally appearing before your eyes.

What's really cool about working in this way is that you're effectively creating song section 'clips', and they behave in a very similar way to any other clip. With the Selection (arrow) tool you can drag them, duplicate them (holding down the Alt key as you drag), or select them and drag their left or right edge handles to adjust the Block start and end times. Making massive changes to Song structure is as easy as rearranging a few building Blocks — hence the name! You can also reassign Blocks: each clip has a pop-up menu triangle next to its name. Click that and you can switch that clip to play another Block.

Blocks-ing Clever

What's maybe a little less cool is that the contents of your individual tracks are not editable; you can see them, but they're greyed out. It's understandable, of course — Reason/Record is referencing the separate Blocks and merely displaying the contents in Song View. But what if you need to edit something, or make localised changes? The great news is that you have loads of options.

Firstly, there's nothing to stop you clicking the Block View button once again, and going into an individual Block to change it in some way. When you click back to Song View, those changes will be incorporated into the song, any time that Block is used. It's very much a 'live' link



In Song View you use the Pencil tool and the new Blocks pop-up menu to literally draw song sections in the Blocks track.

Spot Mute Trick

Using the Mute tool to silence individual track lanes of a Block in Song View is a very effective arrangement device, but it's an all-or-nothing situation, and Reason/Record won't let you mute just a bit of a block. However, there's a fabulously easy workaround. In Song View, take the pencil tool, and simply draw a new clip where you'd like your 'selective mute' to occur. Because this is linear data (albeit empty!) it takes priority over the block data and for that moment, nothing happens!

Blocking Your Old Songs

Blocks is a beautifully simple concept when you build brand-new songs with it. But what if you're mid-way through a project that is built in a linear fashion, and you'd now like to 'convert' to a Blocks workflow? Simple answer: cut and paste. Make sure Blocks is enabled and that you're in Song View. Start by dragging a selection over everything you'd like to represent a Chorus, say. Hit Command-C (Mac) or Ctrl-C (Windows) to copy it, and click into Block View. Paste this into an empty Block, adjust its length, and rename it as appropriate. Back in Song view, delete what you just copied, and immediately draw the block equivalent into the Blocks track with the pencil tool. Carry on like this for each section until the entire linear arrangement is represented by blocks.

between Blocks and your song.

Secondly, in Song view you can use the Mute tool ('T' key) to silence individual track clips within a Block: just point and click, and they become greyed out. This is really handy, allowing you (for example) to create just one fully worked 'Verse' Block, but mute different tracks in it every time it's used in the Song. That can provide a great sense of variation, and saves you having to create and maintain lots of separate 'Verse' Blocks that just have minor differences between them.

Thirdly, you can freely record and add non-block material anywhere you like in Song view. It really is just like the normal, linear sequencer of old, except that it's got that extra Block track. This makes the system hugely flexible. And thank goodness, because it's pretty much a necessity when adding vocals in Record. Think about it — you might use the same 'Verse' Block three times, but if you'd recorded a vocal as part of it, you'd get the same lyrics three times too. Not usually what you want! So the solution is to use the Block/Song system to lay out your backing track, but then in Song View create a vocal track and record on to it in conventional linear fashion 'on top of' the Blocks. Doing this requires no special consideration, and no extra steps — just work as normal.

This sort of 'linear overlay' can also be used to create little links between sections/Blocks, or to add tiny localised variations. You can record right on top of Block material, in any track, and linear song-mode clips always have priority over Block material. To give an example, if I needed a simple, one-bar song intro ahead of a 'Verse 1' Block, maybe a drum fill and a bass lead, I almost certainly wouldn't bother trying to create it in a separate Block. What's the point? It would only happen once, and doing it in Song mode would allow me to hear much more easily how it integrates with the verse.

The final word in seizing back editing flexibility is to convert Block data into old-fashioned linear data. You can do this in Song View. Just select a Block clip, right-click, and choose 'Convert Block Clips to Song Clips' from the contextual menu. Any unmuted clips represented by that Block are transformed into linear, editable copies. In fact, though, they're just overlaid on top of the referenced Block material, so nothing gets deleted. If you want to convert an entire arrangement from Blocks to fully editable linear data, right-click in the track list and choose 'Convert Block Track to Song Clips'. The Blocks content is still retained, but Blocks-style replay is automatically disabled (via the Blocks track's little 'On' button), and you'll be back in a completely linear world once more.

Super Conductors

Pro Tools' rulers not only show the passage of time, they allow you to control it!

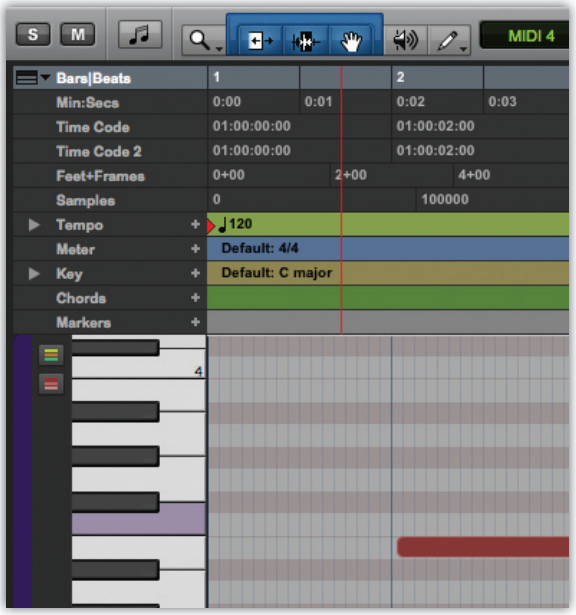
MIKE THORNTON

In this month's Pro Tools workshop, we're going to take a look in detail at the Rulers section of the Edit window. There are a lot of different ruler options, which can be displayed in both the Edit and MIDI Editor windows, as you can see in the screenshots. You can determine which rulers Pro Tools displays by clicking on the Rulers icon, which is displayed in the main timebase ruler. A drop-down menu lets you select which rulers are visible. At the bottom of this menu are two shortcuts: All, which will display all the rulers, and Main, which will display just the main timebase ruler. You can also do a similar thing using the Ruler option in the View menu.

All the different rulers in Pro Tools fall into one of two groups: Timebase or Conductor. Timebase rulers, as the name suggests, measure time, and do so in a range of different formats, from conventional minutes and seconds, through feet & frames for film, or timecode for video, to bars and beats for music. Timebase rulers can measure time in two different ways: sample-based time (absolute time) and tick-based time (relative time). The Minutes:Seconds ruler, for instance, measures absolute time, while the Bars|Beats ruler measures relative time. In other words, the amount of time represented by each division of the Bars|Beats ruler is not fixed but depends on the overall tempo and other factors.

Pro Tools lets you set any track timebase as either sample-based or tick-based. Audio tracks in Pro Tools are sample-based by default. This means that audio regions and events are located at specific sample locations, and they do not move from those locations if the tempo changes in the Session — which, in turn, means that changing the tempo will result in their bar and beat locations changing. On the other hand, MIDI and Instrument tracks are tick-based by default. This means that MIDI regions and events are located at a specific bar and beat locations, and they do not move from that location if the tempo changes in the Session, even though their sample location will change.

You can change tracks from sample-based to tick-based in the track's 'header section'. The clock icon indicates a sample-based track, while the metronome icon indicates a tick-based track.



Pro Tools rulers: the complete collection.

Employing A Conductor

Conductor rulers display a variety of events that can affect or control the timebase. These include Markers (memory locations), Tempo and the Tempo Editor, Meter, Keys and the Key Signature Staff, as well as Chords.

- The Meter (how many beats in a bar) and Tempo (how many beats per minute) rulers indicate changes in meter and tempo within the Session. Tempo and meter events affect the timing of tick-based tracks, and also provide the tempo and meter map for the Bar:Beat grid and the Click track. You can edit tempo events in either the Tempo ruler or Tempo Editor.
- The Key Signature ruler indicates changes of key, which affect MIDI notes when applying transposition. You can edit key and key changes in the Key Signature ruler.
- The Chord Symbols ruler shows chord markers.
- Finally, the Markers ruler displays markers to important track locations.

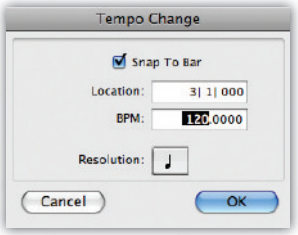
Unlike the Timebase rulers, some of these Conductor rulers are new and still relatively unfamiliar to many Pro Tools users, so the rest of this workshop will focus on working with them. When doing so, I recommend that you change the Linearity Display mode to Linear Tick (relative) scale. This will mean that, irrespective of the tempo, each bar will be displayed as equally long (which will mean that time-ruler scales such as samples or minutes and seconds will change when there are tempo changes).

The Tempo ruler displays tempo, with Tempo Event Markers indicating any changes. You can edit the tempo visually in the Graphic Tempo Editor, or make precise changes in tempo using the Tempo Operations window.

The Song Start Marker is a special tempo event that identifies the start of the song. By default, Pro Tools places it at the start of the Session but it doesn't have to be there. You can move it by dragging it with the mouse, but if you already have tick-based regions in your Session, they will all move too. If you want to move the Start Song Marker and not have the regions move, hold down Start-Shift (Windows) or Ctrl-Shift (Mac) before dragging it, but be aware that it will only move in whole bars when you do this. You can set the initial song tempo by double-clicking the Song Start Marker and entering the revised tempo into the dialogue box that appears.

To insert a new tempo event, click in the tempo ruler at the point where you want the new tempo to start, then click on the Add Tempo button (+) in the tempo ruler header. Then, in the Tempo Change window, you can adjust the exact location

and enter the new tempo, as well as defining what musical value to use for the Beats Per Minute (bpm) setting; the default is a crochet (quarter note). You can also choose to have the tempo event snap to bar, to get it tight on the start of a bar.



The Tempo Change dialogue provides a simple means of inserting a tempo change at a specific place.

Changing Things Around

All existing tempo events can be moved, edited, copied and pasted.

- You can drag tempo events with the mouse. The cursor changes to a 'pointy finger' when you approach an event; to edit it, double-click it and make changes in the Tempo Change window.
- You can delete an event by holding down Alt (Windows) or Option (Mac) once the cursor has changed to the 'pointy finger', then clicking the event.
- To copy and paste a series of tempo events — for instance, to copy a tempo change from the first chorus and paste it on all the other choruses — highlight the range of events in the Tempo ruler, copy them into the clipboard, then place the cursor on the Tempo ruler in the appropriate place and hit Ctrl+V (Mac: Command+V).
- If you hold down Alt (Windows) or Option (Mac) when you make the initial selection, it will extend the selection across all the Conductor rulers.

To access the Graphic Tempo Editor, click on the Expand triangle in the Tempo ruler header section. You now have access to a line that looks similar to the volume automation graph, which is used to graphically edit the tempo through the song. Tempo changes that occur over a number of bars, such as a rallentando at the end of a song, are actually made up of a series of straight lines. A setting called Tempo Edit Density determines how often Pro Tools will put in a tempo event to create your curve. It is accessed by clicking on the Dens: button in the Graphic Tempo Editor header section.

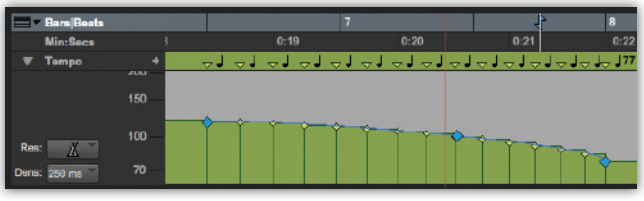
As well as editing, copying, pasting, moving and deleting events using the Grabber tool, you can scale up or down a group of events with the Trim tool and use the Pencil tool either to add new events or overwrite existing ones. As ever, you can choose the most appropriate Pencil tool mode from the drop-down menu — the Line and Parabolic settings are the most useful in this context.

Once you have drawn the line, you will end up with a curve in which the start and end events, together with one part-way through, are larger blue diamonds. These are the curve adjustment handles, and allow you to move the start and finish points and also the final shape of the curve. These update as you drag unless you hold down Start (Windows) or Ctrl (Mac) while adjusting the curve, in which case they'll only update afterwards.

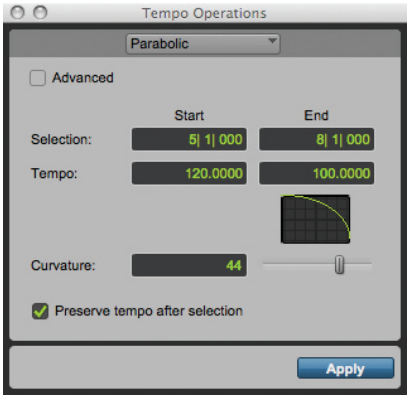
You can select individual tempo events by double-clicking on the horizontal line, and an entire curve by triple-clicking on a horizontal line. Note, though, that if you have the Smart tool enabled, you will need to click in the green area just below the line. If you are on the line instead, the Smart tool changes to the Trimmer.

You can use Shift-Tab or Shift-Alt-Tab to extend the selection forwards or backwards to the next or previous tempo event. This is useful when you then want to use the Trim tool to scale a tempo-event selection.

The Tempo Operation Window is the other way of creating a range of tempo events over time, using a dialogue box to determine what parameters you want to adjust rather than doing it graphically. To open a Tempo Operation window, go into the Events menu, select Tempo Operations, and then either select the Operations window or choose one of the predefined windows from the submenu. For example, I could choose Parabolic to create a rallentando, enter the appropriate start and end points, together with their respective tempos, then adjust the curve and set whether the end tempo reverts to the start tempo (Preserve Tempo) or not.



A rallentando drawn as a curve into the Graphical Tempo Editor. Note the blue diamonds that control its shape.



When you need a complex tempo change that conforms to precise specifications, the Tempo Operations dialogue is your friend.

Other Rulers

The Meter ruler allows you to set and change the time signature (meter) through the song. You can use exactly the same techniques as I have described for the Tempo ruler for graphically inserting, moving, copying, pasting and deleted meter events directly on the Meter ruler. Similarly, if you double-click the meter display on the Transport window, you can use the Meter Change window to create meter events.

The Key Signature ruler will display key signature events, and is useful for when you export your MIDI data to a score-writing package such as Sibelius. You can also apply changes in key signature to transpose data on what Pro Tools calls 'Pitched Tracks', namely MIDI and Instrument tracks. However, you may not want all potential pitched tracks to be transposed when you insert a key signature. (A typical example would be a MIDI drum track.) You can enable or disable this from the track's Playlist menu.

The process of changing key signatures is similar to the other rulers but there is one trap you need to watch out for when you get to the Key Change dialogue window. If you want the Pitched tracks to be transposed, you must remember to tick the Edit Pitched Tracks box, otherwise the change of key will have no affect. Leaving it unticked is useful if you just want to insert key-change symbols into the Session, so that when you export the data to a score package they are already there for you.

Now we have a working understanding of the Conductor rulers, next month we are going to look at creating tempo maps for a variety of sources — including Sessions recorded without a click track... SOS

Don't panic if your iDevice has recently gone to a watery grave; your music may still be salvageable.

MIKE WATKINSON

It seems that a common reason for people to visit a 'genius' at the Apple store is a faulty iPod, often as a result of water damage. This kind of damage invalidates any warranty and, unfortunately, the problem also afflicts iPod Touches and iPhones. Obviously, the smart thing to do is not get it wet in the first place but, if it does get damaged in this way, all is not lost. Maybe your iDevice won't function anymore, but you may still be able to rescue the data. This is especially relevant if you manually sync your music and other files to your device, as iTunes on your Mac will not have a mirror copy of what is on the iPod. So, resist the kind offer from the Apple Store's genius to wipe your device and perform a full system restore, and go home and try the procedures explained here instead.

Back To The Mac

Just to remind us that they are a computer company as well as a manufacturer of mobile devices, Apple recently hosted the 'Back To The Mac' event in San Francisco and took the opportunity to make several important product announcements. Perhaps most importantly for musicians, GarageBand has undergone significant tweaking.

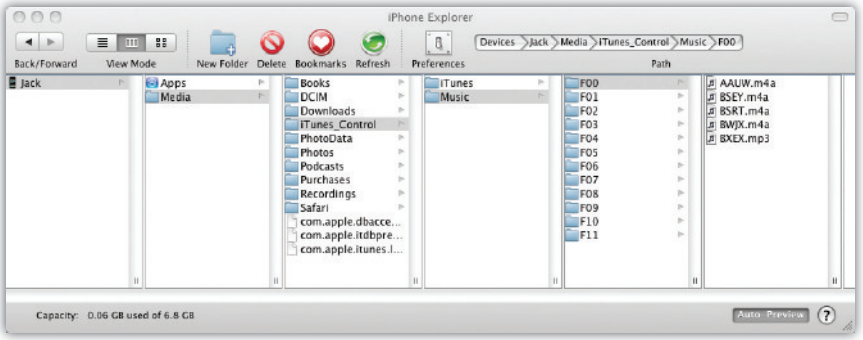
GarageBand 11 gains Groove Matching, FlexTime, more amps and effects, new piano and guitar lessons and a 'How did I play?' feature for measuring your own progress when learning the keyboard. We'll look more closely at GarageBand 11 in a future issue.

Perhaps the major software announcement was Mac OS 10.7 — codenamed Lion — presented as OS X with a user interface layer borrowing much from the iPad iOS.

Also, opening before the end of January will be the Mac App Store: much like the mobile App Store, but for Mac applications — tell me you didn't see this one coming!

Lastly, not one, but two new versions of the MacBook Air were announced, with 13.3-inch and 11.6-inch screens, and internals that owe a lot to the design of the iPad. The world's most desirable notebook just became a size zero...

The MacBook Air was the major hardware update announcement at the recent 'Back To The Mac' event. It's apparently part of the 'next generation of MacBooks'...



If your iPod Touch or iPhone has suffered water damage, don't despair: iPhone Explorer allows you to drag and drop files from your device while it is connected via USB.

Pod Rescue

As long as the device shows up in iTunes, you can select it and, under the Summary tab, ensure that 'Enable disk use' is checked (this will already be the case if you have chosen to 'Manually manage music and videos'). Now, when you check the content of the iPod in Finder you will see that the Music folder is conspicuous by its absence. No doubt there are copyright-based reasons for Apple hiding this folder, but you can make it reappear by entering the following commands into Terminal:

- defaults write com.apple.finder AppleShowAllFiles TRUE [press Return]
- killall Finder [press Return]

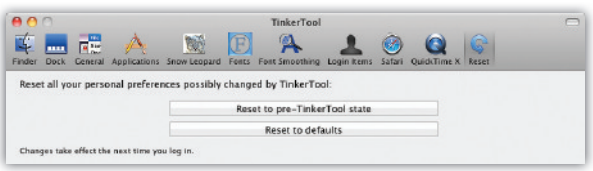
The contents of the iPod in the Finder will now reveal a grey (previously hidden) folder called 'iPod_Control'. Inside this there is a folder called 'Music' that contains several folders beginning with the letter F. In these folders reside your music files. They will have strange names, but don't despair because, once returned to the iTunes Library, you will see that they still contain relevant information. You should drag and drop these files into a suitable temporary folder before dragging and dropping into iTunes. To return the Finder to its standard state, type the following into Terminal:

- defaults write com.apple.finder AppleShowAllFiles FALSE [press Return]
- killall Finder [press Return]

In their infinite wisdom, Apple don't allow iPhones or iPod Touches to be used in disk mode when connected via USB. There are, however, a few Mac applications that will allow you to browse your device when connected in this way, and iPhone Explorer from Macroplant (http://www.macroplant.com/iphoneexplorer) allows you to drag and drop files from your iDevice to the Finder once you have located them.

Terminal Velocity

If typing commands into Terminal is something you consider best left to the specialists in Apple T-shirts, you will be glad to know that this functionality is part of a free-to-download application called TinkerTool by Marcel Bresink (available from http://www.bresink.com/osx/TinkerTool.html). In TinkerTools' Finder Options, you simply tick the box to 'Show hidden and system files' and then click the 'Relaunch Finder' button. TinkerTool also contains many other potentially useful options, such as a tick box to revert the iTunes 10 title bar to standard layout with horizontal buttons (the Terminal commands for which are discussed in last month's column).



TinkerTool has the capability to show hidden files (along with other options) on your Mac without you having to use the Terminal application. Once you're done, you can easily reset it to the 'pre-TinkerTool' state.

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If your laptop's giving you ground-loop problems, maybe it's time to go back to battery power...

MARTIN WALKER

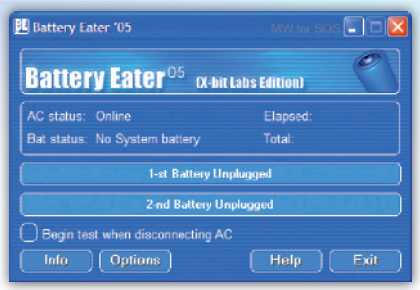
More and more musicians are relying on PC laptops. But, sadly, more and more are also running into ground-loop problems when they plug in their laptop mains power supply, connect the output of their audio interface to other earthed mains gear and then hear various unwanted 'digital nasties' in the background.

I've offered loads of advice in the past on how to solve such issues, and the dedicated thread I created on this topic on the *SOS* forum at www.soundonsound.com/forum/showflat.php?Cat=&Number=222392 has already been read in excess of 70,000 times. However, sometimes the easiest solution for ground-loop problems is simply to power your PC laptop from its battery.

Batteries Included

Up until a few years ago, most laptops had NiMH batteries that, for longest life, always benefitted from being fully drained before recharging. Just to confuse us, the Lithium ones found in today's laptops should be treated in exactly the opposite way.

For the longest life, you should not fully discharge them before recharging, because if you leave them fully discharged for any length of time, they may not subsequently be capable of holding any charge at all.



Want to measure how long your laptop battery will survive when running audio software? Try the Battery Eater bench-test utility.

So if you rarely power your laptop from its battery, partly drain it of power, then remove and store it. Your Lithium batteries should not be constantly trickle-charged from the mains, either, since their charge capacity can apparently drop by up to 20 percent each year when used in this manner, and they can also catch fire if overcharged or damaged. The best approach for longer Lithium battery life is somewhere between these two extremes: use your laptop regularly on battery until it's getting low on power, then charge it up again.

Some musicians are reluctant to rely on battery power, as they are worried that the battery's life will slowly dwindle to nothing, However, there's an easy way to determine how long your battery will currently last under various load conditions. Battery Eater (www.batteryeater.com) is a Windows-only donationware utility that can measure the operation time of your laptop battery under maximum workload (which is, after all, how most musicians use their laptops when running audio), as well as when idle and in less stressful word-processing modes.

Still In The Fan Club?

Meanwhile, for those of us still relying on mains-powered desktop PCs, Seasonic have recently released some new PSU models worthy of a close look. Seasonic PSUs are already extremely popular among musicians for their rock-solid performance and reliability. I've been relying on one in my current PC for almost four years now, and have been well pleased with the very low acoustic noise levels of its fan.

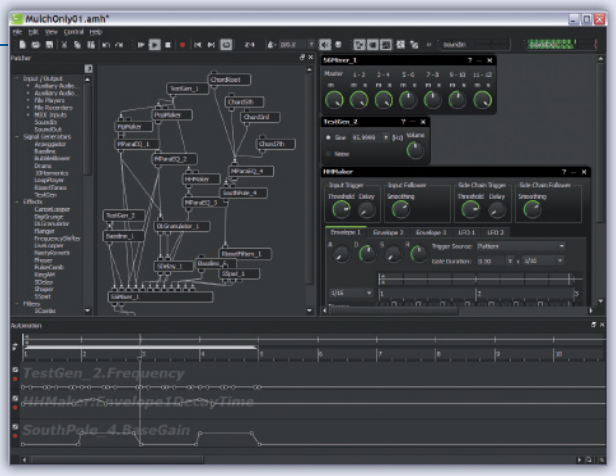
Having said that, the Holy Grail in a studio environment is being able to abandon the fan altogether, so it's not surprising that Seasonic's X-series fanless models are already getting a lot of attention from both commercial and home DAW builders. To design a 'noiseless' supply that doesn't overheat is tricky, but Seasonic have achieved this goal in two main ways, by first offering excellent design efficiency, to minimise the heat generated by the PSU itself and, second, by using a honeycomb-mesh PSU case to significantly increase ventilation, and thus let warm air from inside your computer and PSU escape more easily.

The PSUs are Initially available in X400FL (400W) and X460FL (460W) versions, but these wattages are probably not sufficient for the musician whose PC is stuffed to the gills with DSP cards and RAID hard drives. However, they are excellent choices for the more typical i5/i7-based audio PC, as long as you can afford to pay the premium price: the 400W model typically retails at around £115, which is roughly double the price of a fan-cooled device of similar capacity. However, these units are built like tanks, and they do say silence is golden!

PC Snippets

Front-panel USB 3: While USB 2 audio interfaces now seem to be popping up all over the place, those who are backing up huge files to external hard drives will no doubt be pleased to hear that ASRock is the first motherboard manufacturer to launch motherboards offering front-panel USB 3 connectors. Featuring an Intel-designed 19-pin plug/socket and typically better than 120MB/second sustained transfer rate, this product finally means that users won't have to ferret about behind their PC to plug in an external USB 3 drive. Let's hope other motherboard manufacturers follow suit shortly! See www.asrock.com for more.

AudioMulch 2.1.0: Ross Bencina's incredibly flexible AudioMulch software for real-time sound creation and improvisational composition has been updated to version 2.1.0. Time signatures can now be entered in the various looping modules, rhythmic units chosen in many other modules, and there are also various new rhythmic matrix pattern editors, all of which augur well for polyrhythmic exploration. Along with new dynamics modules, MIDI device hot-plugging and various bug fixes, this latest version is a must for the sonic experimenter and is free to existing users. Check it out at www.audiomulch.com.



The latest version of AudioMulch offers many new features to tempt the sonic experimenter.

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Win! Celemony 10+10 Birthday Box

Bavarian software developers Celemony celebrate their 10th birthday this year, and to mark the occasion, they've teamed up with *SOS* to offer a fantastic competition prize!

Their first product (released in 2001, a year after Celemony was founded), was a little piece of software that allowed you to independently adjust the pitch and timing of recorded audio, provided the source was monophonic. That software was called Melodyne, and it proved to be a revelation. Although the plug-in version of Auto-Tune, which was released a few years earlier, was already capable of automatic pitch correction, Melodyne allowed the user to adjust timing, as well as shifting the formant content of processed audio to make the results sound more natural.

In our review of the original Melodyne, we described it as "a truly revolutionary creative musical tool" — but Celemony's greatest coup was yet to come...

At the Musikmesse show in 2008, Celemony presented Melodyne DNA (Direct Note Access) to the world. Once again, this allowed the independent manipulation of pitch and timing — only this time, you could edit individual notes within *polyphonic* audio! It's fair to say that Melodyne DNA was the most eagerly anticipated audio product in recent memory, and although it took a while to come to market, when it finally arrived, no one was disappointed. Jaws dropped and minds boggled at the possibilities that Melodyne DNA afforded: suddenly, tasks such as fixing an out-of-tune string on a recorded guitar part, or changing a recorded piano piece from major to minor, became possible, where previously they had been the stuff of dreams.

In celebration of their 10th birthday, Celemony are kindly giving away one of the legendary wooden boxes in which the first copies of Melodyne were delivered. And with very special contents: the boxes contain not only the latest versions of Melodyne Studio and Melodyne Editor, but also a voucher for all products and updates released by Celemony for the next 10 years! Given their track record of innovation, who's to say what they'll come up with in that period?

For a chance to win this invaluable prize, simply fill out the form below, answering all the questions and providing your contact details, and send the form to us. Alternatively, you can enter online via the *SOS* web site. The closing date for entries is 11th January 2010.

Prizes kindly donated by Celemony
T 2Twenty2 +44 (0)8452 994222
W www.2twenty2.com
W www.celemony.com

QUESTIONS

In what year was the first version of Melodyne released?
a. 1999 ☐
b. 2000 ☐
c. 2001 ☐
d. 2008 ☐

Where are Celemony from?
a. Bolton ☐
b. Bratislava ☐
c. Bologne ☐
d. Bavaria ☐

What do the letters 'DNA' (as in 'Melodyne DNA') stand for?'
a. Direct Note Access ☐
b. Delete Note Aberrations ☐
c. Don't Need Ability ☐
d. Dipsomaniacs Need Alcohol ☐

Celemony Tie-breaker

'DNA' normally stands for Deoxyribonucleic Acid, which contains genetic information that can be rearranged to create superheroes. If you could become a superhero, what would you do with your superpowers? Answers in 30 words or fewer, please.

.....

.....

.....

Name: .....
Address: .....
.....
Daytime tel. no: .....
Email: .....

If you would like to receive more information about Celemony products, please tick or cross this box. ☐

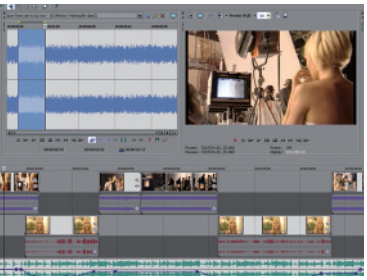
Post your completed entry to: Celemony Competition December 2010, Sound On Sound, Media House, Trafalgar Way, Bar Hill, Cambridge CB23 8SQ, England.

the small print

1. Only one entry per person is permitted. 2. Employees of SOS Publications Ltd, Celemony, and their immediate families and business associates are ineligible for entry. 3. No cash alternative is available in lieu of the stated prize. 4. The competition organisers reserve the right to change the specification of the prize offered. 5. The judges' decision is final and legally binding, and no correspondence will be entered into. 6. No other correspondence is to be included with competition entries. 7. Please ensure that you give your DAYTIME telephone number on your entry form. 8. Prize winners must be prepared to make themselves available in the event that the competition organisers wish to make a personal presentation.

SOUND ON SOUND

VIDEOMEDIA



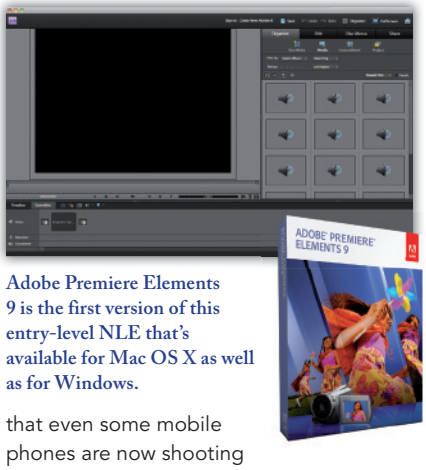
In Focus

Julian G Harding
Video Media Editor

Since a few big video trade shows have come and gone recently, there haven't been quite so many hardware announcements in the world of video and post-production this month — but there's still a lot of news to catch up with on the software front.

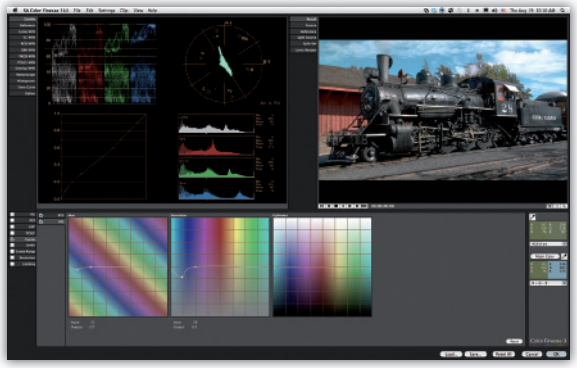
Mac users can rejoice at a new mid-level NLE, in the form of Adobe Premiere Elements 9. Before v9, Premiere Elements was only available for PC, but from now on, OS X users will be able to use this cut-down editing system. Elements has a greatly simplified interface compared to Premiere Pro, with many settings residing 'under the hood'.

Some great features at this price include native AVCHD and H264 editing, allowing you to drag footage straight from a Flip camcorder or DSLR, for example, and edit with no transcoding. The video engine has been optimised for HD resolutions, a sensible move given that even some mobile phones are now shooting video at 720p. Just as Adobe's Encore disc-authoring software has shifted focus from burning traditional DVDs to 'web DVDs', Premiere Elements 9 now allows the same type of export, encouraging users to export their DVD as a Flash element for web sites. Premiere Elements costs £79 from www.adobe.com.



Adobe Premiere Elements 9 is the first version of this entry-level NLE that's available for Mac OS X as well as for Windows.

There are plenty of new plug-in releases, including a new version of Synthetic Aperture's (www.synthetic-ap.com) popular colour-grading plug-in and stand-alone application, Color Finesse, which has been updated to v3. Color Finesse was already a very powerful piece of software prior to v3, and a version is included with Adobe's Production Premium software package. The update includes extended host support, with 32- and 64-bit plug-ins for Adobe CS3 to CS5 and a new native FxPlug version for Final Cut Pro.



Color Finesse 3 introduces some new touches to the already comprehensive grading programme.

For those who aren't used to the depths and complexities of colour grading (the 'Colour By Numbers' feature in *SOS* September 2010 would be a good starting point!), Synthetic Aperture have introduced new Auto Colour and Auto Exposure controls to enable single-click correction, or a neutral starting point from which to try out your ideas. These ideas will be even easier to implement now, as some new colour controls have been introduced. The Vibrance control gives colour a well-tailored brightness and saturation boost, and is a nice addition, as tweaking vibrance generally leads to a subtler, more aesthetically pleasing result than adjusting a standard saturation control.

HSL (Hue, Saturation and Lightness) curves have been introduced as well, allowing control of saturation restricted to a particular colour range, a useful feature which could be used to pick out and enhance all the red parts of a scene, for example. Another nice introduction is the Highlight Recovery tool, which can be used to prevent areas from clipping and turning pure white during colour correction. Just as it is in audio production, digital clipping of colour or lightness information is generally undesirable and results in a permanent loss of detail, so it's good to have a tool to help avoid it. You may see clipping or 'blow out' on people's faces when viewing video shot in bright sunlight; watch out for white patches on the face or forehead!

Version 3 also adds support for the popular Tangent Wave control surface, which includes three trackballs and a great number of other control for quick and accurate colour correcting. If you've not tried using a control surface to colour grade, do find a way to try one out; it makes the whole process much more enjoyable! Color Finesse 3 costs $575 for After Effects, FCP and Premiere Pro, and can be purchased form the Synthetic Aperture web site.

Plug-in powerhouses Waves (www.waves.com) also released two new bundles, both of which bring together selections of plug-ins from their range that will help with post-production tasks. The Waves Video Sound Suite gathers together a useful collection, including the recent W43 Noise Reduction plug-in. It is 'inspired by' the classic Dolby Cat43 hardware unit, and is designed to eliminate unwanted ambient noise, such as wind, recording hiss and even air-conditioning noise, using multi-band reduction. I've often had to jump through hoops trying to solve issues related to location-recorded sound, so the idea of having a single plug-in to carry out exactly these tasks with some degree of automation sounds liberating!

Reverb is taken care of by the IR-L, a lighter version of Waves' IR convolution reverb, which allows the producer to place sounds in lifelike acoustic spaces. The primary use for such processing comes in ADR (Dialogue Replacement), where actors re-record the speech parts in a studio, and these parts are re-synchronised with the video. Such recordings often sound very flat, and a little room

The popular Tangent Wave control surface is supported by Colour Finesse 3, for hands-on colour grading.

ambience can make an overdubbed dialogue sit much more comfortably in the mix. The bundle also includes some other Waves favourites, such as the DeEsser, L1 Limiter, Q10 Equalizer and Renaissance Compressor, all of which are renowned for high-quality processing. The bundle costs $850 direct from the Waves web site.

The second bundle from Waves is the Sound Design Suite, which includes 28 plug-ins (29 if you purchase the TDM version, which includes the PS22 Stereo Maker as well). Notable plug-ins packaged with the Sound Design Suite include Renaissance Bass and LoAir, two powerful sub-harmonic exciters; Doppler, which simulates the pitch-bending effect of an object moving past a listener; and GTR3, a guitar effects suite that is very useful for mangling sounds and creating original timbres in post-production. Downloading the bundle costs $3800 for the TDM version and $2300 for the native version.

The new Waves W43 is a multi-band noise reduction plug-in inspired by the long-discontinued Dolby Cat43 hardware, and comes with the Video Sound Suite.

Waves' LoAir is included with the Sound Designer Suite and enables you to use bass enhancement in 5.1 surround configurations.

This image shows noisy footage on the right, and the result of Magic Bullet Denoiser on the left. Note the smoother skin tone.

Red Giant (www.redgiantsoftware.com), creators of the Looks and Colorista grading programs, have released a new Magic Bullet plug-in for After Effects and Final Cut Pro. Denoiser is designed to remove the characteristic and unattractive grain that is captured by digital sensors when they are set to high ISO or Gain settings. (Much as with an audio amplification system, increasing the sensitivity increases the amount of noise captured.) Denoiser has both a one-click mode, where the software selects the best settings, and a more complex mode that allows fine-tuning of different colour channels, and compensation for both low-speed and high-speed motion. Unlike most competitors, it also offers RGB (red, green and blue) processing, as well as separated colour (chroma) and black & white (luma) processing, which makes it useful for cleaning up digitised film and uncompressed footage. The plug-in is priced at $99 as a direct download from the Red Giant web site.

Perhaps the most exciting recent hardware announcement was the Atomos Ninja (www.atomos.com), which combines a video monitor and hard disk recorder in one unit. When video is recorded, light passes through the lens, hits a sensor or sensors and is converted into digital data. After this, it is usually compressed to fit onto storage media, a process that results in a loss of picture quality. Separate hardware recorders allow massive capacity, and if the camera outputs the video feed at full resolution before compression, they can capture this feed uncompressed, or far less compressed than the camera itself is capable of.

The Ninja can take an HDMI feed from a camera and encode it to any one of Apple's ProRes formats, from the low bit-rate (but still remarkably sharp) ProRes LT to the broadcast quality ProRes HQ. Unlike most portable recorders, it also doubles as a touchscreen monitor that can be used both for accessing menus and achieving accurate focus. What's truly great about the Ninja is that, rather than using expensive proprietary media, it makes use of standard 2.5-inch laptop HDD drives, so you can purchase 500GB for video storage for around £40, or opt for an SSD for peace of mind. That's around the same cost as your average 32GB Class 6 SDHC card, which really puts the price of storage in perspective! Most DSLRs are yet to catch up and offer uncompressed HD output via HDMI while recording, but plenty of other cameras do offer this capability, and the Ninja looks to be quite a deal at €795.

The Atomos Ninja is both a touchscreen monitor and a hard disk video recorder, capturing in Apple ProRes format to 2.5-inch hard disks.

The consumer market has thrown up some interesting recent products too, both of which point towards future trends in video production and distribution. Toshiba announced their first 3D TV (as yet unnamed) that requires no glasses. The full 3D effect can only be viewed from a narrow range of angles, but it's certainly an interesting development. Lastly, Sony have decided to integrate Google TV into a range of new Internet TVs, allowing users to stream straight from the Internet via the television and integrated browser. Both of these developments mean you'll soon have even more ways to get your content into people's living rooms!

Sony Vegas Pro 10

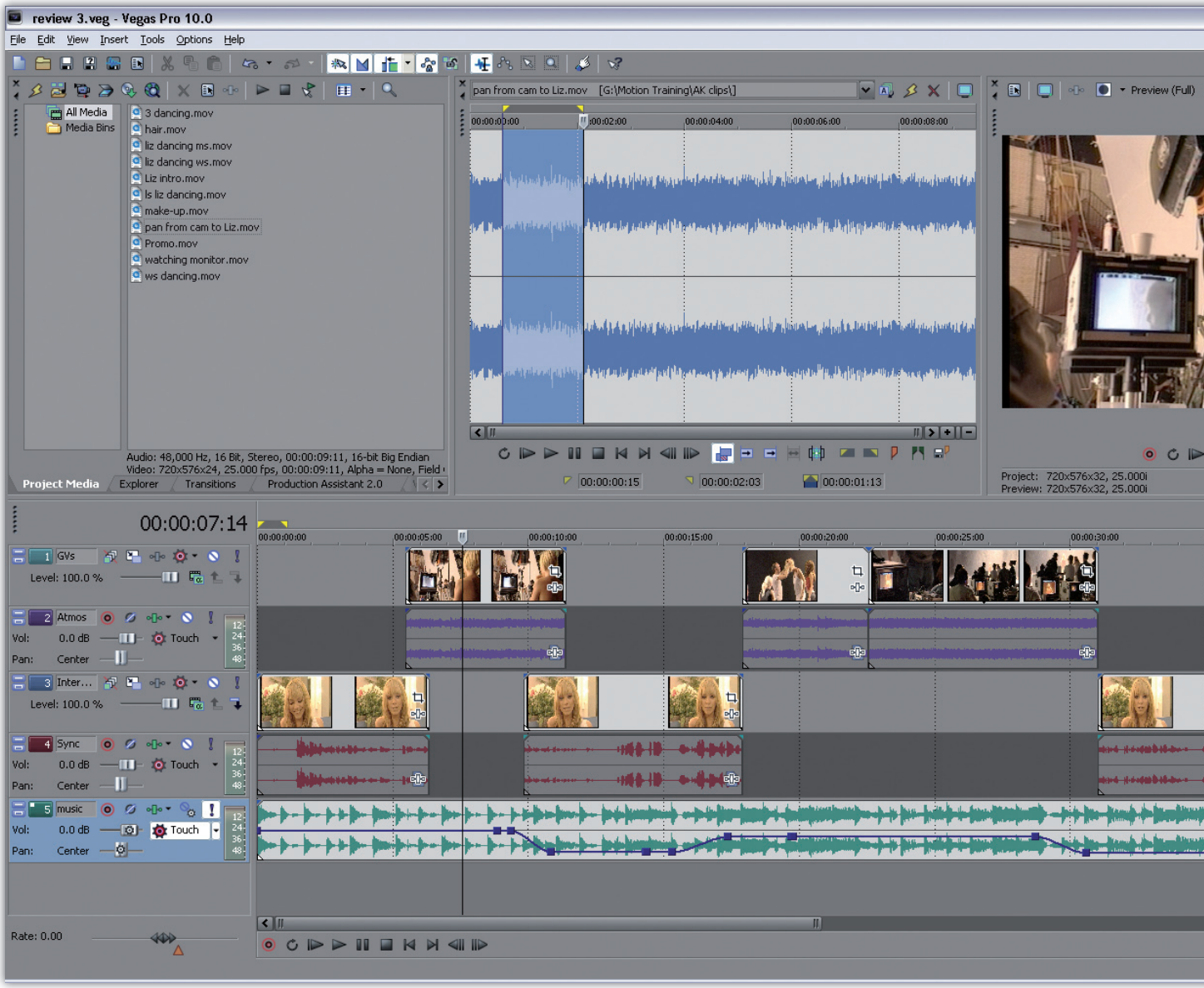
Non-linear Editor For PC

MIKE BUTLER

The latest version of Vegas offers plenty of professional features at a competitive price. So can you expect compromises in performance, or is Vegas Pro a spectacular deal?

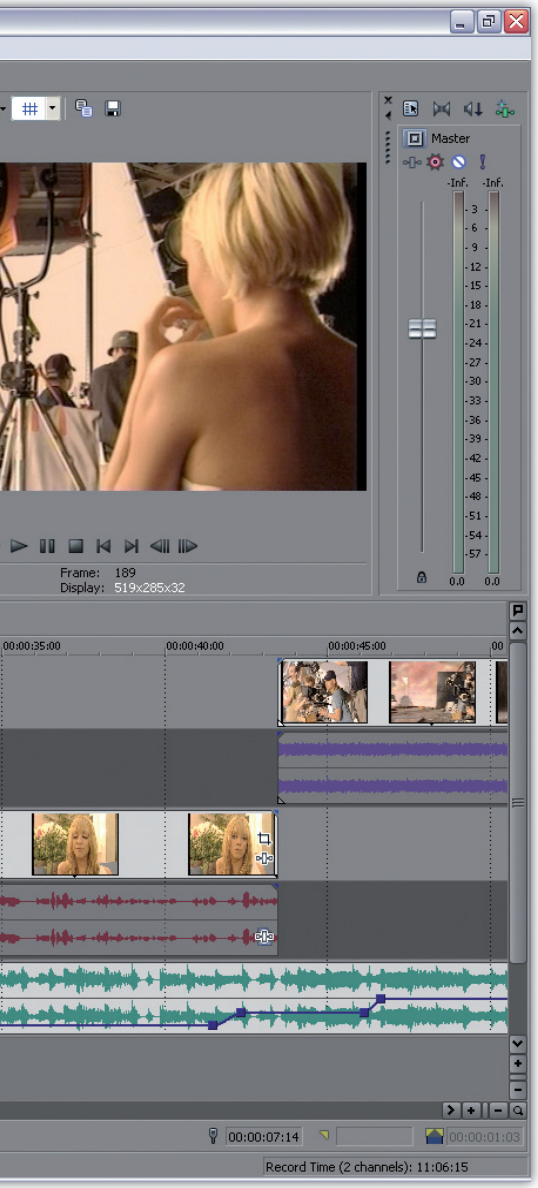
When the chance came to review Sony Vegas Pro 10, I jumped at it. Sony Vegas has always had a reputation for being a little bit 'different' to the mainstream video-editing products, and I was keen to find out why.

Sony Vegas Pro is the professional version of Sony's consumer software package, Vegas Movie Studio. As such it has a whole raft of professional features, including support for XDCAM and Red One media, the ability to interface with the Aja Xena series of video-capture



hardware, 5.1 surround sound mixing and so on. At £581 including VAT, it's relatively affordable as full-featured software goes, being significantly less expensive than Adobe's Premiere Pro CS5, and much less expensive than Avid's Media Composer 5. Included in the package is DVD Architect 5.2 for burning video to optical media, but if you want to add the companion Production Assistant 2 (see the 'Production Assistant 2' box), you'll need to shell out a further £124.

In writing recent reviews of Avid's Media Composer 5 (See *SOS Video Media* September 2010) and Adobe's Premiere Pro CS5 (See *SOS Video Media* November 2010) it's apparent how similar these products, along with Apple's Final Cut Pro (FCP), are becoming. There seems to be something of a convergence of approach taking place. Does Sony Vegas 10 fit this trend as well?



The Interface

On starting up the software, the answer seems to be 'partly'. Unlike FCP, Premiere or Avid, which are both very much studies in grey, the interface of Vegas Pro is alive with colour, populated with multi-coloured icons which are much more pictorial in their design than the rather 'runic' approach taken by most other NLEs.

Compared with previous versions, however, the v10 interface does look more like those of its rivals. The Video Preview window now has transport controls beneath it, the audio meters have shuffled over to the far right, and the Trimmer (used for editing footage before placing it in the timeline) is now displayed as part of the interface by default.

There's a Project window much like that in many other NLEs, where you can organise the clips you're working with, and you can create Bins to separate them into different categories such as stills, interviews, music and the like. Bins can only be viewed in the Project Window — they can't be opened separately as they can in Premiere or FCP — and this means that you can't display the contents of more than one Bin simultaneously, which is a pity.

Unlike its rivals, the default layout of Vegas Pro doesn't have a conventional source viewer: if you double-click a clip in the Project window, it loads straight to the timeline, rather than giving you a preview in a separate window. This was a bit shocking at first, as I'm used to clips loading into a Viewer or Source Monitor, from which I can select what I'm going to edit into my timeline.

The Vegas approach encourages you to you trim off or delete the unwanted material once you've put a clip in the timeline, leaving behind the shot you want. This is all very well if your clips are short and well organised, but it's an approach that falls down if you have long clips, or clips which contain many useful shots.

Fortunately (for me at least!), there's a program preference to change this. Clips will then load into the 'Trimmer', which functions much like other programs' Source Viewers. From there, clips can be marked up and either saved as Subclips back to the Project window until needed, or added to the timeline by dragging and dropping. It feels odd, though, to have to delve into Preferences to do something that seems the natural way to operate.

There are two fundamental kinds of timeline editing. Avid Media Composer is designed to let you build a timeline by selecting the clips you want and adding only the parts of them that you choose: the timeline is additive, if you will. Vegas builds the timeline the other way round, by asking that you put everything in and then remove the unwanted — it's subtractive editing. FCP and Premiere sit somewhere in the middle — they are designed to work equally well either way round, so users can choose their own method.

The Vegas Pro interface is quite reliant on drag-and-drop techniques for many aspects of editing in the timeline. Working in the professional arena, I come across many instances of hard-working editors suffering from RSI (repetitive strain injury) induced by excessive mouse use, and there's a general feeling that doing as much manipulation via keyboard commands as possible is ultimately better for health when using software a great deal. Not only that, but a well-designed keyboard interface is usually quicker to work with. While drag-and-drop is simple and intuitive, it's not really very 'Pro'.

That said, once you get digging Vegas Pro 10 is actually stuffed with professional features, so ultimately whether or not you enjoy the interface comes down to personal preference and method.

64-bit And Acceleration

Both Windows Vista 64-bit and Windows 7 64-bit are both supported by the new version, as well as 32-bit varieties of XP, Vista and 7. Vegas Pro 10 doesn't make such »

Sony Vegas Pro 10 £581

PROS

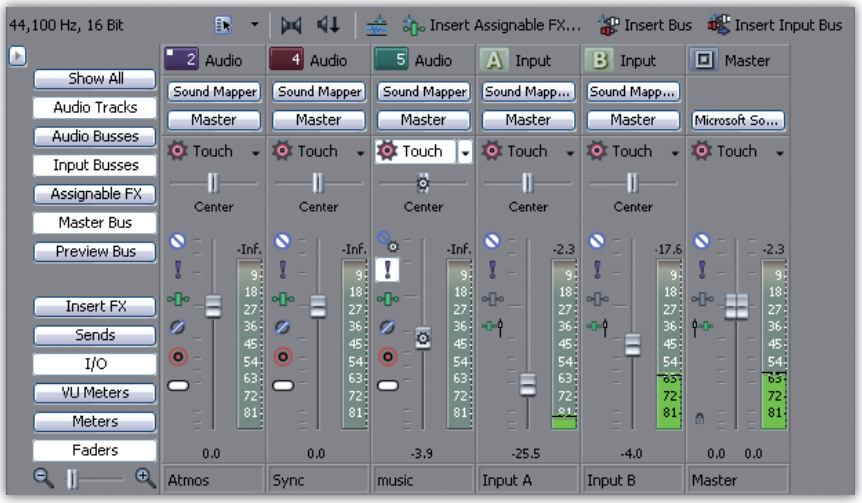
- Relatively low cost.
- 3D editing now supported.
- Comprehensive audio editing and mixer.
- 64-bit with good format support.

CONS

- Simplistic video editing for the price.
- Outdated video effects.
- Lack of in-timeline acceleration.

SUMMARY

Vegas Pro 10 for Windows provides a lower-cost alternative to its main rivals, with some great audio functionality that can't be found in other NLEs. 3D, 64-bit support and accelerated export are all welcome, but it would be nice to see CUDA acceleration in the timeline and keyframing in all effects, both of which would help Vegas Pro catch up with its rivals.



The audio mixer in Vegas Pro is much more comprehensive than that of most NLEs, and now includes up to 26 input buses.

» a song and dance about its 64-bit support as Premiere Pro CS5 does, but you should expect to benefit from the increased amount of RAM available in HD projects. Of course, there's no Mac OS X version available — Vegas Pro is Windows only, so Mac fans like me are out of luck.

Vegas Pro 10 also makes use of Nvidia's CUDA-enabled graphics cards to speed up encoding of projects into Sony AVC format. It doesn't, however, use CUDA to provide the enhancement of timeline playback that we've seen in Premiere Pro CS5, which is a shame. Indeed, the lack of GPU acceleration in Vegas Pro places something of a limitation on the software generally.

Audio Improvements

SOS readers will be glad to hear that there are plenty of nice audio features in Vegas Pro 10! The heritage of Vegas Pro is as an audio editing package, and v10 still has features that show these origins.

The track headers themselves now include audio level meters and pan faders, a welcome feature for those familiar with audio editing.

New in this release are input buses, which allow you to both loop your timeline out to external effects hardware, mix external audio sources into the timeline and monitor external audio sources such as talkback mics. This is something that's not usually so easy to do in an NLE.

You can add up to 26 input buses, which should be plenty for most people. Input buses are added directly to the Audio Mixing Console, and can be assigned to any available input to the PC. Similarly, their outputs can be assigned to the timeline Master bus (if it's to be used to add audio into the timeline) or directly to an output from the PC, if it's not.

It's all very well being able to loop out to an external effects processor, but to make that useful, you need to be able to capture the result. Vegas Pro 10 includes the ability to carry out a 'real time audio render', so that the audio in the timeline is sent out to the effects processor in real time, the result brought back into the system, and incorporated into the final mix.

Audio effects were already pretty well catered for beforehand, but Vegas Pro 10 has added the ability to apply them to individual events on the timeline. Conversely, one of the new features of Media Composer 5 was to allow audio effects to be applied to entire tracks rather than to individual clips, such is the difference in focus of the two.

Audio level monitoring has also been improved. Each track now has a mini level monitor in its header, making it easy to see at a glance what's going on, and there's a proper VU meter display per track in the Audio Mixing Console.

Other audio department tweaks include a dedicated pan fader in each track header in the timeline, and support for .bwf (broadcast wave format) metadata in recorded audio files. All very nice, and in keeping with Vegas' audio credentials. The only surprise is that input buses haven't been incorporated before now!

Video Features

Here's the headline-grabber! Vegas Pro 10 can edit stereoscopic 3D material. In this respect, it's right up there with Avid's Media Composer, which has had 3D editing capabilities for a little while now. It's good to see this capability in a much less expensive software package like this. Of course, the amount of 3D production being carried out at the moment is pretty small, so to some extent this is more window-dressing than real mainstream capability, but the software does a competent job. We can expect to see the 3D production arena expanding in the years ahead.

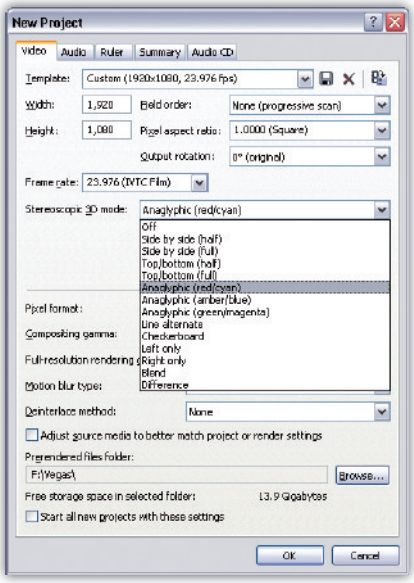
The software can handle either separately shot video from two cameras, or 'combined stream' 3D video from cameras like the new Panasonic HDC SDT750. If shot separately, the left and right images must be combined into a single stereoscopic clip. This is done be first syncing the two up and then combining them via a new command, 'Pair as Stereoscopic 3D Subclip'.

The Project Properties allow you to choose between the different methods of viewing 3D, including anaglyphic mode (using coloured glasses), half-res side-by side mode and full-res side-by side, for example. Of course, you're going to need 3D monitoring to be able to see the results of some of 3D modes, but anaglyphic mode and a pair of coloured specs is reasonably successful.

There's a new 3D effect to allow you to tweak the registration of the two 3D images and adjust 3D depth. This is essential, because viewers find drastic changes in 3D environment from shot to shot disturbing. The Cineform Neo3D codec is also

Alternatives

Full-featured alternatives include **Adobe's Premier Pro CS5**, and the **Production Premium** bundle it is included with, both of which benefit from an accelerated timeline but are more expensive than Vegas Pro. **Avid's Media Composer** is a great piece of software, but will set you back a lot more money, while **Grass Valley Edius 6** is closer in price terms.



Comprehensive 3D project options allow you to choose from a wide range of formats, including anaglyphic 3D, for which you'll need special coloured glasses but only a standard monitor.

supported, although it's not included in the package. Neo 3D has established itself as a worthy 3D format, and Vegas Pro 10 can natively play back, edit and render projects to Neo 3D QuickTime or AVI.

Oddly, however, DVD Architect has not yet been upgraded to support the new Blu-ray 3D format, so your HD project will have to be output in either anaglyphic mode or in side-by-side mode, either of which halves the effective resolution.

Closed Captioning

Closed captioning, or subtitling, has become an integral part of programme-making in these days of multi-language broadcasting. Closed captions are subtitles that you can choose to display, as opposed to open captions, which are subtitles that are 'burned into' the video and can't be switched off. Generally, closed-captioning information is encoded on line 21 of the SD video frame, just before the beginning of the actual picture on line 23. Increasingly, there's a need for NLEs to be able to preserve any line 21 data through the capture/edit/export process, and Vegas Pro 10 has fallen into line with this.

The software can also import caption files from a variety of formats and marry them up to the timeline, and there's support for displaying the captions in the Preview window.

It's also possible to create captions directly in the timeline, although you'd be a bit of a masochist to do so, such is the complexity of this apparently simple task! Formatting of the caption text is by Caption Markup commands, similar in style to HTML tags. Since there's no nice interface to help with this, however, you have to hand-code the commands into the dialogue box. For this reason I'd say it's better to use some external captioning software for the task.

Timelines with closed captions can be exported for Windows Media Player, QuickTime Player, YouTube and Real Media Player, as well as to DVD Architect, the bundled DVD Authoring package.

Format Support

New HD video formats appear at a dizzying rate these days, which is keeping the NLE software writers on their toes. Vegas Pro 9 already offered support for various AVCHD-based formats including DSLR media and Sony's own NXCAM.

Working with file-based media of this sort can be a mixed blessing in some respects — for example, every clip has the same timecode, generally starting at 00:00:00:00, making logging on-set a nonsense. Sony's NXCAM format has the ability to attach time-of-day timecode information to the AVCHD media, and Vegas Pro 10 can read and display this.

Improvements have been made to the handling of QuickTime-based AVC media, such as that produced by DSLRs. This has been a particular beef with Vegas Pro 9, and will be welcomed by existing users. Regarding Project formats themselves, Vegas Pro 10 newly supports 720p/50 and 720p/60 formats for capturing, output to tape and rendering to AVI, which is a useful addition.

Video Effects

The big noise in the video side is of course the 3D editing, but some work has been done in the video effects department as well, although not to the extent that many users might have been hoping. Unfortunately the lack of 'keyframing' (the video equivalent of automation) for effects parameters is still a frustration in v10, and you cannot easily automate changes over time.

A solution could be on the way, however. Vegas Pro 10 introduces support for plug-ins written to the specification of the Open Effects Association. An SDK is available, and Sony hope that the new capabilities »

Here you can see two 3D modes: red-cyan anaglyphic mode (left) and half-res side-by-side mode (right).

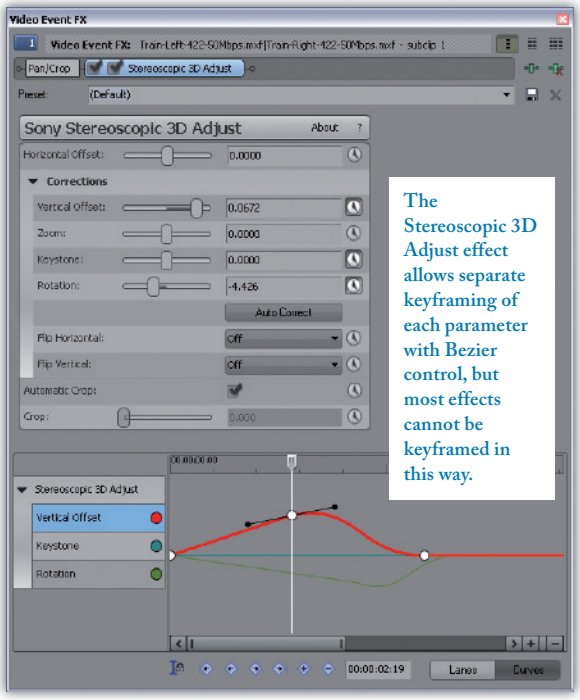
Production Assistant Pro 2

Production Assistant Pro isn't a standard part of the Vegas Pro package, but it's worthy of mention here because it's a little gem that any owner of Vegas should consider purchasing. Production Assistant functions both as a stand-alone piece of software and as an 'extension' to Vegas Pro, accessed from the View/Extensions menu.

As a stand-alone program it can be used to set up and create Vegas projects from user-defined templates, saving the user the hassle of defining format, file locations and so on. In a pressurised production environment, this sort of hand-holding can be invaluable to avoid mistakes being made by staff under pressure (yes folks, I always did hate editing against a deadline!).

As an Extension within Vegas Pro, Production Assistant can perform useful tasks such as batch-processing media, which is useful for fixing issues such as video aspect ratio, audio normalisation and channel assignment, where you have lots of files.

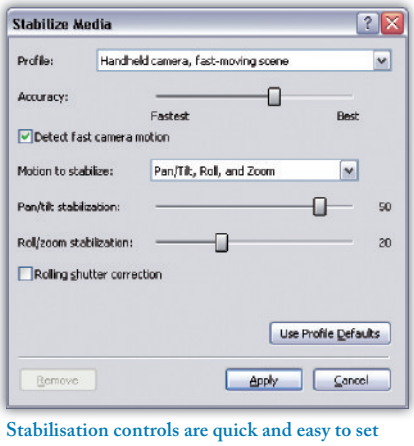
Importantly, it does these in the background while the editor gets on with the editing job itself. There's also a useful facility to automatically scan for memory cards, and upon discovering footage from card-based cameras, copy it to the project's media folder without interrupting workflow. This is similar to the functionality of FCP's Log and Transfer tool and Avid's AMA, and gives Vegas Pro a real boost in the competition with its rivals.



The Stereoscopic 3D Adjust effect allows separate keyframing of each parameter with Bezier control, but most effects cannot be keyframed in this way.

» this offers software developers will result in a new generation of video effects becoming available. The first fruits of this are seen in the 3D adjustment effect, with each parameter separately 'keyframable' and a choice of interpolation methods between keyframes. These are facilities that other NLEs have long taken for granted, and Vegas Pro needs to play catch-up in this aspect, as the current system is not as simple to use as that in Premiere Pro, for example.

Also new in Vegas Pro 10 is an image stabiliser, which as its name implies, tries to remove the wobbles from handheld camera footage. Oddly, this isn't applied as an effect, but is instead a menu item, available when right-clicking an event in the timeline. By comparison with other image stabilisers I've used it was quick to render, but don't get your hopes up too much. In my test clips the results were only a modest improvement in stability, even when set to maximum quality, so stable camerawork is still the best option.



Stabilisation controls are quick and easy to set up, but don't expect the effect to work miracles on very shaky footage!

Multi-camera Editing

Multi-camera editing is one of those features that is likely to appeal to *SOS* readers, since it's a good technique for the production of music videos. Once you've shot the performance using two or more cameras, you can sync them up and then edit 'live' on the timeline, switching back and forth as the sequence plays back as if you were mixing different cameras live.

Vegas Pro 10 has some useful improvements to its Multi-camera Editing mode, and after editing you can expand the single multicam track into multiple tracks again, in order to choose whether to keep the inactive angles in the timeline, or delete them altogether. This makes both exploring other possibilities and re-editing the multi-camera sequence much easier. Additionally, it's possible to view the output full-frame on an external monitor, without needing to leave the Multi-camera Editing mode first, which is quite an improvement.

Other smaller improvements include: revisions to the Device Explorer that allow drag-and-drop importing of clips from memory card-based cameras directly to the Trimmer or the timeline; revisions to the interface with the AJA Xena card to allow genlocking (clocking) of the system to an external reference; and improvements to the Print to Tape dialogue to allow users control over the location of pre-rendered files.

DVD Architect Pro 5.2

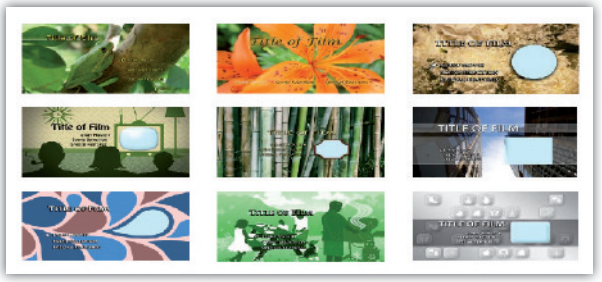
The bundled DVD Architect Pro application has also been updated. Buyers of Vegas Pro 10 will get version 5.2 of DVD Architect Pro. Changes are not exactly extensive — there are a number of new Themes (with preset layouts for menus and the like), and the software has been revised to use the Microsoft IMAPI drivers for disc burning in place of Sony's drivers. This has been implemented to improve disc burning reliability, with more supported disc burners.

Otherwise, nothing else has changed. As mentioned previously, it would have made sense to have included support for the new 3D Blu-ray format, given the capabilities of the editing package, but this has not been done, so DVD Architect Pro remains much as before.

Conclusion

Sony Vegas Pro 10 continues to snap at the heels of its mainstream rivals even though it only pulls away from them in its 'home territory': audio editing. From a video point of view it splits editors down the middle, with fiercely loyal supporters and equally vehement critics.

Offering support for many of the current broadcast formats at this price point makes Sony Vegas Pro 10 a reasonable budget choice. However, performance is adequate rather than spectacular: though it's the lowest-priced, it's also the least powerful of the professional NLEs.



DVD Architect Pro 5.2 includes a number of new Themes and is a solid piece of software, but not too much is new.

As an upgrade from the previous version, I'm left with the feeling that more could have been done to enhance key areas such as video effects and video playback performance. Hopefully these will receive a boost in the next release!

Still, for those who are used to the DAW environment and looking to learn video editing, Vegas Pro could present an easier and less expensive way to transfer ediitng and software skills than some of its fuller-featured rivals.

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Flip Ultra HD

Pocket Video Recorder

The Flip Ultra HD promises high-definition video recording in the palm of your hand — but does it live up to the claim?

PAUL WHITE

A few months back, I decided to dip my toe into the murky waters of video recording, by buying a Flip Ultra HD pocket camera. I bought my particular 8GB Ultra HD model back in May, though models have been updated a little since then! The 8GB model can record two hours of 720p HD video and operates for the same amount of time from its lithium-ion battery pack, when fully charged. The battery is charged via a pop-out USB connector, but if you do find yourself running out of power mid-project you can switch the battery pack for a pair of standard AA batteries, and carry on shooting. There's no expansion port for a memory card, so two hours shooting is a maximum before dumping to a computer via USB.

Carrying lots of separate bits and bobs is annoying, so a fully-functioning camera that slips into the pocket like this is a very attractive proposition. In case you're worried about keeping it safe, you also get a thin fabric bag with the camera. I've lost that already, so I use an iPod sock!

The latest version of the Ultra HD records at 1280 x 720-pixel resolution, and models from September 20th 2010 onwards all use H264 video compression and AAC audio compression, and employ the MP4 file format. The newest 8GB, two-hour model (but not the cheaper 4GB model) also includes image stabilisation and a feature called FlipPort, which lets you plug in external accessories, such as a wireless mic. My own model doesn't have this refinement.

All the Flip cameras also include the required video player and codec installers, stored in the camera's memory rather than on an external CD-ROM installer, which is extremely convenient.

Make It Simple

The camera is very simple to operate, with a fixed-focus lens, a big red button to start and stop shooting, and a four-way cursor pad. A power button on the side of the camera turns it on or off, and you're all ready for shooting within a couple of seconds of powering up. The four-way pad can be used to operate the 4x digital zoom during filming. The zoom travels at a fixed rate and, being electronic rather than optical, results in a loss of resolution as the zoom level is increased. Two further buttons are used for replaying clips and deleting unwanted ones.

Moving backwards or forwards through the clips you've already taken is accomplished using the left and right cursor buttons during playback. A small (around 30 x 42mm) but very clear colour screen shows you what you're shooting and is also used to play back clips. A user preference lets you decide whether the red 'shooting' LED on the front of the camera should light during recording or not, and this little LED doubles as a charging indicator, too.

The accompanying Flipshare software package is used to transfer files from the camera to the computer, and to allow basic editing and assembly of the clips to form a complete movie. It allows basic start and end titling, as well as simple editing of clips.

Once the camera is connected to a computer, the clips are displayed in a window, and you have the option of importing them while also leaving them on the camera, or clearing the camera after importing the clips. They are stored in folders labelled by month, depending on when they were recorded, which is

Flip Ultra HD 8GB £139

PROS

- Inexpensive
- No extra cables or cards needed.
- Basic editing and archive software included.

CONS

- Light weight makes steady handling difficult.
- Built-in sound recording not for serious music use.

SUMMARY

The Flip Ultra HD may be basic, but it can deliver surprisingly good results and is useful when learning basic video techniques. While the included software is great for knocking up quick holiday videos, you'll need something a bit more sophisticated to create professional-looking music videos.

Audio Recording Quality

The built-in mic works well enough for 'snapshot' recording, and is also fine for sync line-up purposes when capturing main audio to a separate recorder. Unfortunately, it has an automatic gain control that can't be defeated, so it doesn't deliver particularly good results for music. It does, however, handle a wide range of decibel levels pretty well, from basic dialogue to a live band as heard from the audience. If you put the Flip too near the loudspeakers, it gives up and overloads, but that's hardly surprising. One potential complication is that the audio is recorded at 44.1kHz, whereas 48kHz is the standard for video work, so you may have to jump through a few conversion hoops if you want to work in a more professional video editing program. Be sure to check!

very easy to grasp, and also helps to keep things nicely organised.

Editing Software

Once open in Flipshare, individual clips can be trimmed at the start and end and then saved over the original file, or can be renamed and saved without affecting the originals. When editing, you can decide whether to use the sound recorded via the camera, a mix of this sound plus an imported MP3 track, or just an imported MP3 track.

Editing is very basic, and comprises dragging the clips into the project window in the order you'd like them to run in, though you can re-order them by dragging again, if necessary. You're given the option to insert start titles and end credits, and the only transition available is a fixed-time, fast crossfade that feels fairly comfortable. Most importantly, the resolution of the edited video doesn't seem any worse than that of the individual clips, which is not always the case when importing into a third-party editing program. For example, you can drag clips directly into iMovie but the quality suffers very noticeably.

Your finished FlipShare video can then be saved as a new file, with further options to post it online as a movie greeting card or online-hosted movie, or burned to a video DVD. You can also extract stills from your video clips at decent snapshot quality.

When I first bought the camera, the software's clip transitions were very messy, but with the latest update they've become much smoother. It's worth updating Flipshare when updates become available, and the software kindly informs you of these when your computer is online.

Practicalities

While the camera is technically HD, the very basic optics and low-cost electronics mean that footage isn't quite as sharp as that of dedicated HD video cameras. Still, at laptop screen sizes the picture quality is certainly good enough for YouTube and other on-line services, especially where music is the main focus of the video. I was also impressed by how well the camera performed indoors at domestic lighting levels, though footage is slightly less detailed than when shot in bright daylight.

The fixed-focus lens means that there's no annoying auto-focus hunting, and the picture remains sharp at distances down to around half a metre. Of course, this system precludes any sophisticated depth-of field tricks (to put the background out of focus, for example) or other tricks such as shifting focus between one object and another, but at least it's one less thing for the novice to worry about.

The biggest challenge turns out to be keeping the camera steady, as its small size and light weight lend it no stabilising inertia. This is particularly noticeable when starting and stopping recording and when operating the zoom. There's a standard tripod thread in the base of the camera, so attaching a flexible tripod (such as a GorillaPod) will help. I also cannibalised an old lightweight tripod to make a monopod, which yields a great improvement over working hand-held. When used outdoors, the built-in mic is quite susceptible to wind noise, but other than that, it works well enough for routine tasks such as close-up dialogue recording.

While the included Flipshare software allows you to edit a perfectly respectable movie from consecutive clips, it is less than ideal for music use. There's no timeline, and as such no way to perform insert edits and break up a full-length, fixed-camera shot of a musical performance. You also have to edit the length of any accompanying MP3 audio that you add, because if it is longer than the video, it will just stop dead at the end of the movie. An auto-fade-out option would be useful for knocking-up simple holiday videos, or other visuals that run alongside music but don't have to be synchronised with it.

As with most cameras, the LCD screen is hard to see in bright, outdoor light, but it's fine for indoor use. I also found that while the camera charged perfectly from my Macbook Pro, the two mains-powered USB chargers I have didn't seem to work too well, with the camera never becoming fully charged. It seems a touch fussy in that department! The matte, rubberised coating is also wearing off the case in places, so protection may be a good investment.

Alternatives

There are plenty of alternative pocket video recorders available, having different resolutions, strengths and weaknesses. **Zoom's Q3** includes high-headroom and high-quality audio recording, and the new HD version records up to 1080p. For those interested in recording underwater, or even on a racetrack, **GoPro's HD Hero** will give you the ability to make a video in almost any conditions.

Verdict

Despite its technical limitations and very basic editing software, the Flip Ultra HD has proven to be a great introduction to video recording. It lets you learn all the basics of framing shots, panning and editing, before moving on to something more 'grown up'.

Daylight pictures are sharp and colourful, while low-light images are less detailed.

There's no additional memory card to buy and no cables to lose, so everything is very self-contained and simple. Its picture quality exceeded my expectations and is fine for pretty much any online publishing application, though for making more than the most rudimentary music videos, third-party editing software is pretty much essential. Best of all, since it is smaller than most phones, you can keep it in your pocket, so you can capture the unexpected!

£ 8GB £139; 4GB £99. Prices include VAT.
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Make A Mobile Music Video

Shooting & Editing A Music Video With The iPhone 4

Apple's iMovie may be designed for making slideshows of snapshots and holiday videos, but what's to stop you thinking outside the box?

J G HARDING

Shooting a video for your band or solo project need not involve spending a lot of cash on gear or spending a lot of time learning video-specific terminology and techniques. Depending on what you want to achieve, you may now be able to make a promotional video using a mobile.

That sounds far too good to be true, right? Yet video technology has advanced to such a degree in recent years that a number of mobile phones are capable of shooting video at HD resolutions, and some of these handsets are capable of carrying out basic editing and titling.

Having recently upgraded to an Apple iPhone 4, I was eager to explore the possibilities opened up by the 720p video recording, as well as Apple's iOS iMovie App. Since I'm accustomed to using both 'prosumer' and some lower-end professional equipment, creating a project like this was something of a departure from my usual processes. I suspected that giving up full manual control would be a pretty difficult pill to swallow. Once you get used to having control over focus, exposure and lighting, your ideas begin to form in such a context.

Play To Your Strengths

So why on earth would I want to try such a thing when I have nice cameras and lenses at my disposal? Well, for a start there are plenty of times when the opportunity to record a great piece of footage arises and I don't happen to have an HD camera, tripod, shoulder support, lights and all the rest to hand! In fact, I'm only likely to have all of the above if I've planned a shoot meticulously and have a very specific goal in mind. Sometimes, lovely as it is, all that gear is just overkill.

Having a mobile that's capable of high-definition video capture is great for shooting a performance as and when its happens, but it's only now, when such devices are capable of editing the video footage they capture, that we can think of doing more.

While there have already been quite a few high-profile videos created with the iPhone 4, all of those I've seen at the time of writing have been edited using a 'proper' NLE, and heavily colour-graded and post produced, usually with plenty of noise reduction, too. What I was more interested in is how much you can do with just a phone, not importing the footage for editing, and using no noise reduction or colour correction at all.

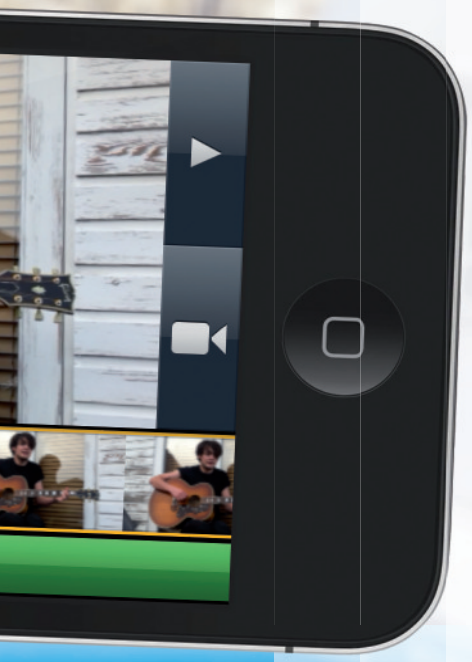
The Making Of...

Having such a tiny sensor (See 'iPhone Camera Vital Statistics' box), the iPhone 4 would not be the best low-light camera, so outdoor shooting would be a wise choice! With no access to multi-layered editing, either. I'd have to rely on a single audio take for the audio, or import a separate file. So I got in touch with singer-songwriter (and bassist for Atticus Black artist Sometimenever) James Page, and asked if he could send me an MP3 of a track that he'd be happy to lip-sync to.

A week later, I arrived at the wonderful Sarm studios in Notting Hill, London, where James was recording. The engineer

See It In Action

To see the video I shot and edited using the iPhone, head to http://www.vimeo.com/16316927



on hand kindly lent us their roof terrace (while Ian Brody lent us a Gibson J200!) for half an hour, so we could shoot a few performance takes and cutaways with nothing but a phone.

The iPhone 4 controls exposure and focus automatically but, cleverly enough, you can touch the screen at any part of the picture and the phone will attempt to focus on and correctly expose that part of the image. Simple as this system is, it can actually be used to pull focus from one object to another and create interesting exposure tricks. It's also possible to create flashing transitions simply by covering the lens for a while and swiftly removing your finger. The camera will fade from pure white to correct exposure, and you can trim the clip to start at pure white for a nice transition into a shot.

James played the track out loud using his own mobile and played and sang along, while I took one continuous take from a still position (resting the phone on the arm of a deck chair) and a few more while moving around and getting some different angles. For each take I made sure not to stop recording. That way I could use the audio to match up with the MP3 I was going to use in the final edit.

The focus range of the iPhone 4 is from around 6cm to infinity, so for cutaways, getting in close will make the most of the lens. The closer you get, the more likely it is that you can create a shallow depth of field. I was actually surprised at the phone's ability to throw the background out of focus when close to a subject, an uncommon trait for such a small sensor. You won't be able to get a softly blurred background when shooting an interview, for example, but your cutaways and close-ups can look rather nice.

Keeping the iPhone still enough while shooting is very hard, not only because the CMOS rolling-shutter effect (also known as the 'jello' effect) exaggerates camera shake in an unnatural way, but because the device is not designed to be held still like a camera. The iPhone's slim profile is great when it comes to fitting the phone in your pocket, but not so good when you want to hold the camera still.

However, by the end of our short session I had come to realise the real benefit of such a portable integrated device: minimal setup time! In the end, I had a tiny window of time in which to capture all the footage I would need, and I was glad not to have to fill lots of it with setup and strip-down time.

iMovie For iOS

Designed for very simple jobs, iMovie is Apple's consumer editing software, included with the iLife package. Despite being targeted at making montages of holiday footage, it can actually be used to chop together a simple edit in a consumer video format. It has recently been ported to iOS for use on iPhone 4 and the fourth generation iPod touch, both of which use the recent and relatively powerful (in mobile terms) Apple A4 processor. The App costs £2.99 from the App store. Currently there's no editing software for the iPad, but this is sure to change, and I believe we'll see something like Final Cut Express appear on future models.

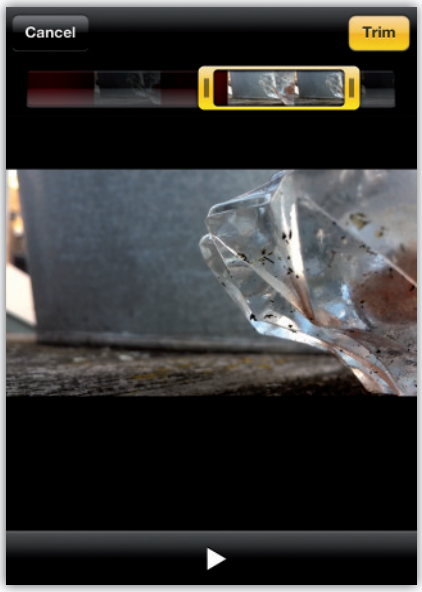
Individual clips can be trimmed before »

iPhone Camera Vital Statistics

The iPhone uses a tiny 1/3.2-inch 5MP CMOS sensor to capture images through a fixed lens. The lens itself has a fixed aperture of f2.8 (which is nice and wide, letting in a good amount of light for its size) and a focal length of about 2.85mm. In photography and cinematography focal lengths refer to how 'zoomed in' the picture is, with higher numbers making the subject larger and closer. Because smaller sensors make images appear cropped (or closer and larger), focal lengths are often expressed in terms of their 35mm sensor (or film) equivalent field-of-view, where a 'standard' focal length, close to the view captured by the naked eye, is 50mm. In these terms, the iPhone lens has a focal length of around 30mm, a relatively wide field of view. This wide field of view helps the phone to focus very close up, as well as capture wide landscapes.

Video clips are recorded at 1280 x 720 pixels, using 24 progressive frames per second, which is the same frame-rate that most films run at. They are encoded to the H264 codec, resulting in a nice-quality image, yet a relatively small file size.

Video recording mode has a simple layout. The point of focus and exposure is controlled by touching the part of the picture where you want it to be.



Opening a video in the standard photo library allows you to trim your clips down very simply.

» they're brought into iMovie, using the standard photo browser, where both in and out points can be set. It's a simple and efficient system: you simply drag the start and end points with your fingertip and press 'Trim'.

Once you're in iMovie, you choose a 'Theme' to start with, which can have background music and titles. I left the music switched off, as I wanted to use the audio track for James's MP3. You can only use either the theme music or an MP3. If your clips have sound as well, iMovie will mix them automatically, ducking the background music. The titles are all a bit 'cheesy', so I didn't use them. The 'Modern' theme is the subtlest, but the option to have simple, plain text fade in and out would be nice.

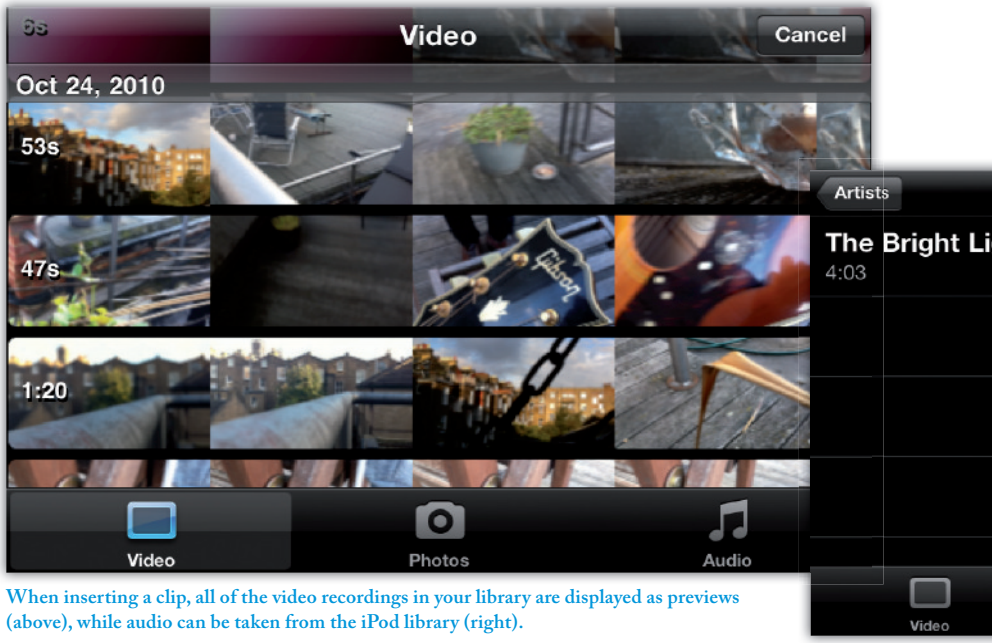
Arranging clips is simple enough. After you choose to insert a video clip, you are presented with a list of all the recorded video clips in your library, complete with preview frames and lengths. Tap the clip and it drops into the timeline in the nearest available point to the current play position. If you tap the clip once to highlight it, you can view the length in minutes, seconds and tens of seconds, and alter that length by dragging the yellow posts at either end. Double-tapping the clip brings up a menu for switching the audio on and off, titling and a location tag. The location tag appears to be of little use, as there aren't a lot of locations listed. The audio switch, however, is very useful for lining up the location audio with the MP3, so expect to be turning this setting on and off quite a lot!

If you decide that a clip isn't right for the edit, you can simply touch and hold it to lift it up, then drag it off the timeline entirely, whereupon it disappears in a puff of smoke.

Editing Limitations

As amazing as it is that you can even think about editing video on a phone, the software has limitations when compared with a traditional NLE. Since there's no timecode to tell you where you are in the sequence, the process of editing is very visual. Also, you can only edit to an accuracy of a tenth of a second, where a true NLE would allow frame-accurate editing (one 24th of a second with this kind of footage), so accuracy is limited.

Pinch-to-zoom (where you touch with two fingers and move them apart to zoom in, and vice-versa to zoom out) helps immensely when making adjustments, and



When inserting a clip, all of the video recordings in your library are displayed as previews (above), while audio can be taken from the iPod library (right).

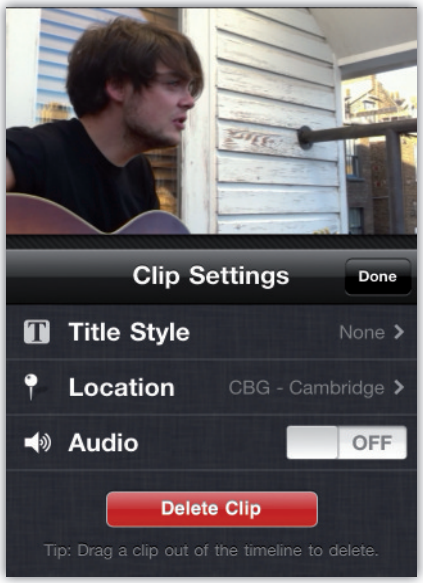
Accessories For Mobile Video

The mobile-phone accessory market is rather large, and any new Apple product, in particular, is followed by a whole raft of cases, portable chargers and other assorted add-ons. It's only now that phones are capable of sufficient processing power to shoot HD video and record multitrack audio that we're seeing the emergence of related accessories.

Joby — makers of the ever-popular Gorillapod flexible tripods — produce a number of different 'Gorillamobile' models, designed specifically for holding mobiles steady during photography and video capture. The tripod legs resemble strings of beads, allowing a Gorillapod to be stood like a traditional tripod, or wrapped around available architecture in inventive ways.

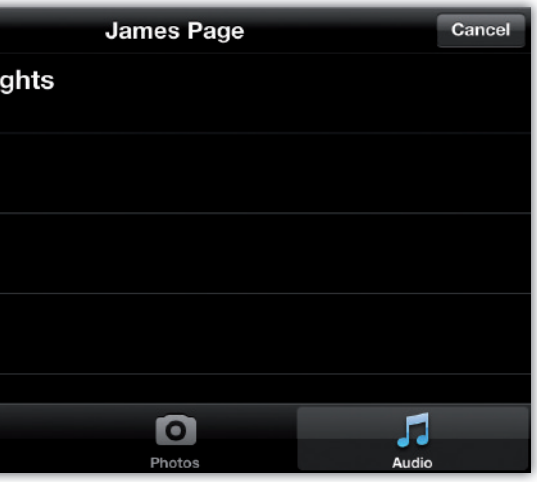
Owle make two models of their Bubo video support for iPhone, with one model designed to fit the 3G/3GS and another for the iPhone 4. Both are machined aluminium supports that resemble a Formula One steering wheel, and have four tripod mounts and a screw fitting for attaching wide-angle adaptors and lenses. The mount has already been used by some inventive types to attach SLR lenses to iPhones for video recording, and looks to be quite a solid solution to the stability issues encountered when shooting with small, light devices. There are also plenty of eBay-only

mobile tripods available cheaply, but don't expect the same high build quality.



Each clip has a few settings that can be adjusted, including titling, audio on or off, and a location setting, the last of which I didn't find a use for!

I found myself zooming in as far as I could to synchronise timing, then zooming back out again to make timeline navigation a little less exhausting! While touchscreen navigation is wonderfuly intuitive, the physically small screen size takes a little



getting used to.

With only a single video track to play with on the timeline, editing is rather linear. Inserting a clip between two others pushes everything further along the timeline, while in a true NLE you would be able to split one clip using another, slipping and rolling footage either way without changing the total length of the piece. For this reason, it's extremely difficult to go back on a decision, as you'll have to re-match your timings for the subsequent clips. Forethought is rewarded here!

You can work around this a little, though. Let's say you have a clip in sync with the MP3 underneath, and you want to put a cutaway clip beforehand. Touch the main clip and note the length, then drop in the cutaway and trim it to the desired size. Touch the cutaway and note its new length, then pull the start of the main clip forward by that many seconds. You'll probably find you have to 'pinch zoom' in and out a lot to do this properly, but it should help you keep all the clips in time.

iMovie doesn't play nice with other programs running in the background, as it places significant strain on the phone's memory and processor. At one point, I found playback very choppy, but closing a few background Apps solved the problem. That said, I was able to complete a large part of both my video edit and this article on a train using just the iPhone 4, with iMovie and Documents To Go Premium writing a Google Doc in the background. I think I've forgotten that these pocket computers make phone calls!

Once your edit is complete, you can export it at three resolutions, the maximum being 720p (1280 x 720), the size of the original footage.

Conclusions

So would I do it again? I'd happily use the phone to make more videos, but I don't think I'd attempt a project in exactly the same way. I enjoyed the challenge and the learning experience, and was quite pleased with the result, but I'd like to try out some of the stabilising devices available. I get the feeling that if you accessorise too much you start to defeat the point of the exercise and may as well use a video camera, so perhaps a mount or tripod would be enough to improve video performance. Also, note that the internal mic is adequate but not amazing, so using it as a sole sound source is not the best idea.

Since making a heavily edited piece is so long winded, it's probably best to either tightly plan how long you want each shot to be or simply not attempt to make a heavily edited piece! You'd be more likely to pull off a successful drama or short film, since you can trim the start and end of a clip and put them in a predetermined sequence, rather than having to keep everything strictly sync'ed to a separate music track.

One key frustration is that if iMovie had two video tracks instead of one, creating a satisfactory edit would be much easier. Perhaps this is beyond the memory and processor capabilities of the device, but with two tracks one could run a performance video on the bottom layer and place cutaways on top, without having to worry about interrupting the synchronisation of the main shot. We'll have to wait and see what Apple come up with next.

I was happy with the final video considering the limitations, but just a few little additions to iMovie would make the whole process a little easier. As it is, the simplicity of the system means that it's necessary to work around serious limitations if you want to create anything other than a simple edit.

It seems to me that dismissing the use of mobile technology for audio and video production is foolish. These devices will soon be powerful, portable computers that simply happen to have a SIMM card slot in the side. Add a large sensor and a lens mount, along with a full NLE, and you eliminate the current distance between camera and edit system.

There are three options for export resolution: two standard-definition settings and a 1280 x 720 HD setting.

Though the iPhone 4 is not the best tool for making a complex edit, it's great for interviews and other simple videos, and perhaps a simple stabilising mount would be good enough to solve most of the camera-shake issues. I think the experience of shooting and editing like this has given me a little glimpse of the future...

5 BEST BUYS

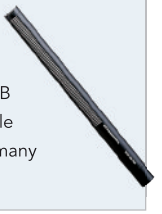
The Gear To Get You Filming Fast

New to the world of video production and unsure what gear to buy? Our guide to the five best options in three categories will get you started.

SHOTGUN MICROPHONES

When shooting anything other than a music video, you're likely to want to capture sound on set. Shotgun mics are highly directional, rejecting sound off-axis and letting you capture just the vital voices. As with all mics, there are models to suit each and every taste and budget.

- **1. Azden SGM X1 (from £99):** A popular low-priced mic, which comes with hot-shoe shockmount and foam windshield. Known for a warm sound when compared to other shotgun mics. Powered by a single AAA battery.
- **2. Audio Technica AT897 (£275):** A short shotgun that can be powered by an AA battery or phantom power. Has a rich but bright tone.
- **3. Rode NTG2 (£199):** A great mic at a good price, the phantom-powered NTG2 is moisture resistant, robust and known for balanced sound and good value.



- **4. Rode VideoMic (£99):** Low-priced short shotgun mic powered by a single 9V battery, and sold in a kit complete with shockmount and foam windshield. It connects using a 3.5mm jack, perfect for small camcorders.
- **5. Sennheiser MKH416 (£766):** Popular premium shotgun mic for those with the money to spare. Rugged and resistant to the elements, it has 130dB peak-SPL handling and a neutral sound, without the brittle high frequencies that can mar cheaper models. Used in many movie productions.

PORTABLE AUDIO RECORDERS WITH XLR MIC INPUTS

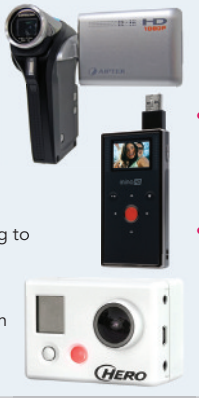
Although some camcorders have built-in microphones, you're usually better off recording to a dedicated audio device. All of these models have XLR inputs with phantom power, so you can use your favourite mics for location sound.

- **1. Edirol R44 (£699):** This four-track unit is solidly built and full featured, including built-in effects on each channel and a pir of mics, as well as four XLR inputs and 24-bit/192kHz operation.
- **2. Fostex FR 2LE (£439):** A two-track portable recorder that records to Compact Flash media, this is well-priced for its build quality and features.

- **3. Marantz PMD661 (£519):** A two-track recorder with simple operation and rugged construction, recording to SDHC media.
- **4. Tascam DR100 (£319):** A two-track SDHC-card recorder with built-in stereo mics, recording up to 96kHz, 24-bit. Solid and simple to use, with Li-Ion, AA and DC power options.
- **5. Zoom H4N (£279):** Zoom's recorder can record from stereo built-in mics and two XLRs at the same time. Records to SDHC card and is rubberised and durable.

POCKET VIDEO RECORDERS

Camcorders and DSLRs aren't the only way to capture digital video. Here are five portable video devices that will slip into your pocket and let you capture footage out and about. These are ideal for those wanting an easy way to shoot with no need to learn menus and settings.



- **1. Aiptek ActionHD GVS (£169):** Mini camcorder from the US with flip-out screen and 5X optical zoom, shooting 1440 x 1080p and 720p to SDHC card. Records 30fps only, but a low price for the features.
- **2. Flip Mino HD (£89):** Popular pocket-sized 720p video recorder, runs for around two hours on a fixed internal battery. Recording at 30fps, shooting to 8GB internal memory only. There's a handy USB connector built in.
- **3. GoPro HD Hero (from £199):** Unique action camera shooting at up to 1080p, 30fps. Comes in kits to attach it to helmets, cars and surfboards! Waterproof casing and wide-angle view make it a versatile piece of gear, shooting to SDHC card.

- **4. Kodak Zi6 (£99):** Kodak's portable recorder shoots 720p footage and sports a large 2.5-inch screen. Shoots at up to 60 frames per second to SDHC, allowing for sharp slow-motion video.

- **5. Zoom Q3 HD (from £199):** The only portable video recorder with a clear focus on audio quality, offering stereo 24-bit WAV recording at 48kHz, at SPLs up to 130dB. HD version offers 1080p/30 video.

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TIME+SPACE

Notes From The Deadline

TV Music From The Inside

The only way to make money as a media composer is from publishing — but greedy production companies are targeting our back end with a vengeance.

PAUL FARRER

It might come as a surprise to some, but there really is no money in writing music for TV. Very little. In fact, unless you are extremely efficient, lucky and work at warp speed, the chances are you will spend more money composing and producing music for a project than you will be paid to do so. The only real money exists in having already written music for a project that happens to get broadcast in lots of different places.

So you can see that this job is entirely about the back end and not about the front. In other words, you tend to look at each project as a lottery ticket. And more often than not, they don't pay off.

Publish & Be Damned

A long-time crusade of mine (and of BASCA, the British Academy of Songwriters, Composers and Authors) has been against the business of coercive publishing: the practice whereby a production company or broadcaster demands to be the sole publisher of a musical work as a condition of awarding the commission.

Different production companies tackle the issue of screwing you with your trousers on in different ways. Some offer a 60:40 split in your favour, to tempt you to sign to them. Some offer their services as publisher, and say that if you don't publish with them it's perfectly fine and it will "absolutely in no way affect our future working relationship" (some even manage to say this with a straight face). Some companies are more subtle, dropping veiled hints and suggestive encouragements like "Give us all your publishing on this job, or you don't even get the chance to send us any pitches and you'll never work in this town ever again."

And who is on our side in all this? Well, hardly anyone, I'm afraid to say. The reason is that competition commissions — who dictate the rules for these kind of scenarios both here and in the EU — were set up to defend the rights of the customer from cartels. And unfortunately for us, they appear to view humble, defenceless institutions like BSkyB, ITV and the BBC as the customers, and freelance composers as the powerful, shadowy, greedy, price-fixing cartel.

I can't imagine a more solitary, disconnected and uncoordinated profession than that of a TV composer. Neither can production companies. Which is why they think they can treat us this way and get away with it. And they're right.

Proud To Be A Union Man

While I'm not proposing a radical, unionised, communist, utopian collective, it's obvious that the more divided we are, the easier it is for third parties to dictate the terms of business. Getting involved with BASCA is the most effective method we have to make our case to the lawmakers with one voice. If you aren't a member yet, you should be, because if we are going to be treated as if we are a cartel it would be nice to know that we have at least the potential to be a little more organised than a herd of cats.

Mao Tse-Tung once declared grass to be illegal and was directly responsible for the deaths of up to 70 million of his own people. But in his defence, he did once write a very successful little red book in which he noted "To produce good music, the 10 fingers should move rhythmically and in coordination." Which I think is his way of saying we should unite or die. Either that or he was expressing regret that he was never taken seriously as a jazz pianist.

"Well, boys, we're a cartel now! How does it feel to be part of a global capitalist behemoth?"

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INDEX

vocal care

Q Does freezing really improve the quality of valves?

I've heard that some companies routinely freeze valves and certain electrical components to improve their performance. Would this technique actually have any effect on quality?
Clive Edmondson, via email

Assuming that it's done slowly and in a controlled environment, valves can be frozen in liquid nitrogen for a while without being damaged. Some companies offer this as an option on the valves they sell, as it is said to improve long-term reliability.

***SOS* Technical Editor Hugh Robjohns replies:** There are certainly some people who claim that it can, but I've not experienced it personally and my knowledge of materials science isn't up to giving a definitive view, I'm afraid.

However, the basic idea of 'deep cryogenic treatment' is to freeze the valve to an extremely cold temperature (well below the capability of any domestic freezer: typically, the valve is submerged in liquid nitrogen at about -195 degrees Celsius and stored like that for a day or so before gradually returning it to room temperature. This cryogenic process is claimed to allow the crystalline structure of the metals used in the valve plates to realign in a way that allows electrons to flow more easily (ie. resistance is reduced), and that's what brings the claimed benefits to sound quality.

Certainly, strange things can happen »

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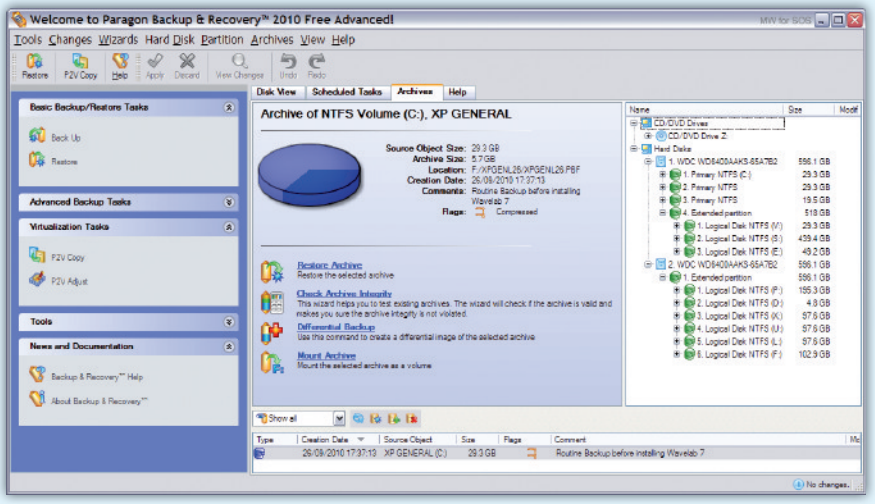
» to metals at very low temperatures — like superconductivity — but these effects don't usually last when the metal is returned to room temperature. NASA also apparently use cryogenic treatments to prepare materials for use in space. But whether the science behind their applications extends to the use of audio valves at room temperature, I'm not so sure. Allegedly, the benefits of the cryogenic process remain throughout the life of the valve, despite the heating and cooling cycles it will go through in normal use.

Given the natural variability in valve sets anyway, I think it likely that differences will be heard between normal and cryogenically treated tubes. However, whether those differences are really better or just different — and whether the cost is justified — I suspect comes down to personal choice.



An imaging utility (such as Paragon Backup & Recovery, shown here) will create a complete snapshot of your computer's hard drive and save you having to reinstall software after a crash or drive failure.

What's the best way to back up my data?

I have a PC running Windows XP, and currently back up my data regularly using copy and paste onto an external hard drive. The problem with my current drive-to-drive method is that, in the case of a full windows reinstall, it means that I still have to reinstall some software (and this eats into my downloads and licences from the developers' web site). Is specialised disk cloning or imaging software a better option?
Paul Allerton, via email

***SOS* contributor Martin Walker replies:**
You're very wise to back up your data regularly, but there's a fundamental difference between copy and paste backups and disk imaging or cloning: the latter takes a snapshot of everything on the chosen partition or drive, including all the hidden and system files, although it's clever enough to ignore such things as huge page files that only contain temporary data, to keep the image file sizes to a minimum.

Each image file is, therefore, a time capsule, since when you 'restore' it your computer will return to exactly the same state it was in when you created that image file, leaving your licensed software intact. Most imaging utilities also offer compression options, typically squashing the data to about half the original size, while still letting you explore and restore individual files contained within.

It's always safest to store these image files onto external media, such as DVD or an external hard drive, to cope with the worst-case scenario of your entire computer blowing up or being struck by lightning. In which case, when you get your replacement PC, you can restore your external backups onto this and avoid days of reinstalling Windows and applications, and authorising copy-protected software. However, if, like many musicians, you have several hard drives in your audio PC, you can also store routine images of one drive to another, so that if one drive goes belly-up, you'll still have a recent image file on your other drive to restore when you've replaced the faulty one.

We explored different strategies for backing up your data in the October 2007 issue of *Sound On Sound* (see www.soundonsound.com/sos/oct07/articles/data_protection.htm), and it might be worth »

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» giving that a read for more information.

There's a bewildering array of imaging utilities on offer, so here's a quick rundown of some of the best alternatives. Windows 7 now includes its own Backup & Restore application that many users find perfectly sufficient, and for Windows 7/XP/Vista a very popular commercial package is Acronis True Image, especially since free licensed versions are available from various hard-drive manufacturers, for use with their own products.

There are also various utilities that work with all makes of hard drive and are totally free for the home user. After trying out quite a few, I've ended up using Paragon Backup & Recovery 2010 (www.paragon-software.com/home/db-express), which has a free advanced version that is fairly unique among the free utilities in offering an incremental backup option that only stores files newer than your previous image files. I've been very pleased with its clear and easy-to-use interface, and my slimmed-down 10GB Windows partition takes just four minutes to back up to a compressed file of some 6GB.

This free version of Paragon Backup & Recovery 2010 runs on both 32-bit and 64-bit versions of Windows 7, Vista, XP and Server 2003/2008. It can create a Linux/DOS recovery environment onto CD, DVD or Flash memory (so that you can boot your PC from these media and restore image files even if your computer refuses to boot up by itself). You can even assign a drive letter to 'mount' an archived partition, so you can access the files within using Windows Explorer. More sophisticated versions are available, but to date I've been perfectly happy with the free one.

Q How can I improve the sound of my DJ set when mastering?

I'm a drum & bass DJ and I work exclusively with vinyl, which is all mastered for a club sound system. When I record sets — for example, for promotional purposes — they sound tight and full through my high-quality system, but through the majority of other systems they sound weak and tend to lose a bit of the tightness in the high end. How can I make sure the overall sound is full and, therefore, more consumer friendly?
Kieren Bagley, via email

***SOS* contributor Simon Langford replies:** Without knowing the specifics of your system it is hard to give any specific advice, but there are a few general points to consider that may help. Firstly, high-quality systems are just that: playing back any music on a high-quality system will almost inevitably sound better than playing it back on cheaper, more consumer-level systems. Instead of comparing how your mix sounds on your high-quality system to how it sounds on a consumer setup, you might be better off comparing how your mix sounds on a consumer system to how other (similar style) mixes sound on that same setup. If there is a commercial mix (one that has been professionally mixed and mastered) that has a similar style of tracks on it, it might provide a better frame of reference for you. This is especially true if there is another mix that you feel does sound good on these consumer systems.

Another thing to consider is that, in mastering your mix, you will be mastering tracks that have already been mastered! So applying additional mastering could suck the life out of the individual tracks. I'm assuming that while you're doing the mix you're adjusting and compensating for any overall level and EQ differences between the different tracks, so that your final mix is pretty consistent in terms of level and tonal balance. If that's the case, I really don't think you need to master your mix at all.

You don't mention how you're recording »

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This is Apple Logic's built-in Match EQ, which allows you to analyse the frequency spectrum of your track and your reference track. It then provides a correction EQ curve with variable depth, to allow you to tailor the amount of correction applied.

» the mix. If you're recording into a PC or Mac, you might be better off reducing the output level from your mixer and then applying some kind of limiter plug-in to the mix once it's been recorded. By doing this, you minimise the risk of overloading the audio inputs when you're recording, and can then bring the overall level back up to maximum. You could, of course, simply normalise the final stereo mix as well. One other thing you might be able to do, if you record the mix into a computer, is to use another commercial mix as a reference (as I mentioned previously), importing that into your DAW. There are a number of plug-ins that will analyse the spectrum of the reference mix and then do the same to your mix, offering up a correction curve that will go some way toward matching the spectral balance of your mix to your reference mix. In my experience, these plug-ins rarely make the target track sound exactly like the source track, but they can help you make broad changes to the EQ that will get you a little closer.

If you're still having problems, it might be worthwhile looking at your equipment. All vinyl records have what is called the RIAA EQ curve applied to them before they are cut to vinyl. This reduces low frequencies by about 20dB and boosts high frequencies by about 20dB. The reasons for this are purely physical ones associated with achieving maximum levels when cutting vinyl records. When a vinyl deck is connected to a DJ mixer, a reverse EQ curve is applied to restore the original sound. I wonder if, perhaps, there's something a little amiss with your mixer or decks if the sound you're getting is consistently different from other mixes you hear. »

duplication

I think it's definitely worthwhile working through a process of elimination to pinpoint exactly where the problem is. The issue you're having — that of your mixes translating to different playback systems — is something that mix engineers and producers the world over have to deal with on a regular basis. There really isn't a mastering technique that can do anything about that. It's just important to constantly reference what you are doing against other mixes on a variety of systems and, eventually, you will learn to factor in the difference in sound and create mixes that sound the best they can on the widest possible variety of setups.

Q What's the best order for recording a band?

I'm going to be recording my band's new album and it's my first big project, so I'd like a bit of advice. It's a rhythm & blues band consisting of drums, rhythm guitar, lead guitar, acoustic guitar, bass guitar and lead and backing vocals. I have a limited number of mics, so I'll be recording sources separately. The question is: in what order do I do this? The band are pretty tight, but I don't think the drummer would like playing to a click track with no reference to the song itself.

I seem to get mixed views on this: I know the normal way is drums first, but what would be best for this band?

Via *SOS* web site

***SOS* Editor In Chief Paul White replies:** It always gives the best feel if the band play together, even if it's only drums plus a rough mix as a guide of the other instruments and a vocal, recorded down one mic, as these can be replaced later. Of course, you also need to minimise spill, so using POD-style devices and headphones for guitars and bass can help in getting the guide parts down and avoids the need for a mic, though you may need a small mixer. Otherwise, put the drums in a different room from the guitar amps. Some people manage by recording just the drums and bass together, but trying to do the drums on their own is asking for problems, as the feel will never be quite right.

If the music is of a type where the drummer is happy to play to a click track, you could always record the separate parts to a click or guide drum loop, then have the drummer put the drums on last. But, again, you could lose all your feel that way.

***SOS* Reviews Editor Matt Houghton adds:** With a genre like rhythm & blues, the feel of the playing is so critical to getting a good result. There are plenty of great recordings where most parts were played with everyone in the room to get a good vibe. As Paul suggests, you need the rhythm section, in particular, to be really hitting the groove, and with that in mind, I would certainly want to track at least the drums and bass together as a starting point. You could have both playing in the same room, using a solid gobo or two to provide separation between the kit and the bass amp, with the amp separated from the kit, but with the bassist standing in a position where he and the drummer have good visual communication. Alternatively, put the bassist in the room, but DI his bass, perhaps running it through an amp

» simulator for monitoring purposes. If you get an amazing take, it would be simple enough to use a good amp simulator such as IK's Ampeg SVX or Softube's Vintage Bass Room, or re-amping, to make it fit in the mix, but you also have the opportunity of laying a new bass part over the drums.

While you could take the 'playing together' principle further, and put guitars, keys, vocalists and whatever else you wish in the same room, you'll often find that achieving acceptable separation becomes problematic, and maybe even a problem that outweighs the benefits of having everyone perform together. In my experience, you're better off having a vocalist and guitarist either in a different room than the drums, or in the control room, with the guitar DI'd and/or run through an amp simulator, so that the whole group is doing a live take into the monitor mix. You can always track those parts, in case you get a moment of magic — with the guitars, as with the bass, amp simulations or re-amping are valid approaches here — but you'll still have the option to overdub those parts later, and many musicians will be glad of the chance to try a few different takes.

Recording the various instruments in a band in the right order is important, as it will affect the feel of the track.

Photo: Richard Ecclestone

What else you record in what order will depend on the other musicians. Does the vocalist want to hear the other parts? Do other musicians take their cue off the vocals? If you make sure you record the guide parts as you go, those questions become less of an issue, and while I often advocate starting with vocals (or other primary elements) when mixing, I don't usually find that it makes such »

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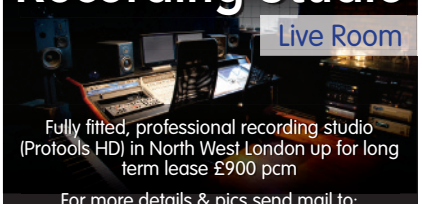

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» a difference when recording.

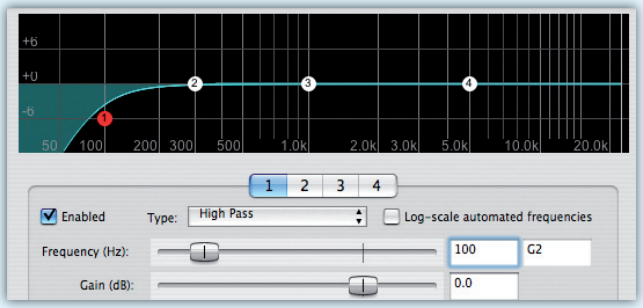
When it comes to tempo and click tracks, your approach will depend very much on the band in question. When you say 'tight', that might mean that the drummer can keep a rock-solid tempo, or it might just mean that all the musicians can keep in good time with each other. In my experience, some bands that sound tight can actually accelerate and slow down considerably during a track, which may or may not be a good thing. The one thing that I would say, though, is that the thresholdsfor what seems acceptable when playing live is different than when listening to a record that gets played again and again. So having a click track or guide track for the drummer might be useful. Really, the best advice I can give is to discuss this with the drummer, go with what they feel comfortable with and simply be alert to any problems.

If you do choose to use a click track, my advice would be to feed it to the drummer alone, and then have everyone else lock in with him or her, which is what a good live band will usually do, after all. And do make sure that the headphones don't leak that sound into the overhead mics, which is something I hear a lot on material sent in to *SOS*!

Of course, it's perfectly possible to overdub drum parts, but I invariably find that when doing this you lose almost all of the magical glue that holds a track together. In this style, more than most, that will probably prove unacceptable. In fact, I can only recall one occasion where I've done it and obtained a satisfactory result.

Q Why does using a high-pass filter make things seem louder?

Why does applying a high-pass filter to a sound sometimes result in the output being noticeably higher than it was before? Today I have been working on a sound that peaks at 0dBFS. It has a lot of low-frequency content. I am applying a high-pass filter at around 100Hz and the output from the EQ is peaking at around +4dBFS. Why should



There are various reasons why using a high-pass filter on a signal can make it sound louder. Some equalisers, for example, actually boost the region just above the turnover frequency, which can produce an increase in peak level.

this happen? Most of the power in this sound is in the low frequencies, and it has little going on above 2kHz, so surely with the high-pass filter most of the energy from the sound has gone!
Via *SOS* web site

***SOS* Technical Editor Hugh Robjohns replies:** This is a very common effect and there are several possible reasons for it. Fundamentally, the filtering process changes the shape of the waveform, so although there may be less total energy in the signal, the peak amplitude may well increase.

If you think about a bunch of different-frequency tones all playing at the same time, their phase relationships vary continuously and add to or cancel each other to create the total waveform. Remove some of those tones and some of those cancellations won't occur. That can result in the waveform becoming bigger.

Most equalisers also introduce significant phase shifts and that, again, will change the way different frequencies combine and cancel. It can also happen because some equalisers actually boost the region just above the turnover point below which they are attenuating, potentially increasing peak level.

The Advertisers Index

Is music meaningless in isolation?

ALEX MARTEN

What is music? The consensus shared by musicians and philosophers of aesthetics alike, is that it is a form of expression. We strum, hit, scrape, blow and tickle (occasionally) things in order to make sounds, rhythms, melodies and harmonies that will express those things that are otherwise inexpressible, or at least hard to express.

Exactly what we try to express through music varies wildly, from adolescent angst, to religious fervour, to streetwise swagger, and all manner of emotions in between, but what all forms of music have in common is that they strive to express *something*.

Expressing yourself is about communicating your inner feelings to others, and this is what I am rather slowly driving at: without others to express our music to, the music very quickly loses its purpose and, ultimately, its value. This is why the creation of music is inherently a social interaction; it is about people coming together to express things to each other and to the rest of the world; it is about *communication*.

And how is this relevant to you, O world-weary *SOS* reader? It's relevant to you because, slowly but surely, musicians and producers like you are being encouraged (no doubt for insidious, shadowy purposes) to make music alone.

It is now surprisingly easy to bypass all forms of direct human interaction when creating music. All the gear you need can be ordered online and delivered to your door. The gear in question can include virtual session players who can replace your friends or, if this isn't enough, you can virtually 'jam' with other musicians around the world via services such as eJamming.

"Without others to express our music to, the music very quickly loses its purpose and, ultimately, its value."

It's only a matter of time before Native Instruments release a VST plug-in called 'Rekord Kontrakt' that not only composes a song, plays all the parts, mixes and masters it, but also submits the resulting track to a record company and generates a meticulously modelled virtual rejection letter from said company, thus completing the process known only too well to many of us *real* musicians. Ultimately, we lowly humans won't be required to engage in the process whatsoever. Hooray! Er...

OK, this may be a *little* bit far-fetched, but it is true to say that every single part of the music production process can now be carried out 'in the box', alone in your room.

But can it really? Yes, the so-called 'Communications Revolution' — the rise of the Internet and (almost) instantaneous data sharing — has meant that we can share information that can be communicated digitally, but does this apply to live music?

I would argue that it doesn't, that no super-high-bandwidth information uber-highway, no cutting-edge online collaboration service, can match the immediacy of the communication involved when performing in a room with other players. There is an almost magical instantaneity to this process, an immediate psychic connection — *no* latency, not just 'low latency' — and, for this reason, there is a great joy to it that cannot be replicated by computer-based interactions, let alone by virtual accompanists. The resulting music has depth, expression and soul.

Many home producers will know the feeling of emptiness that can accompany a day or evening of coming up with perfectly good musical ideas that just don't seem to go anywhere. This is because music is a conversation — making it completely alone is like talking to yourself, and we all know where that leads...

About The Author

Alex Marten is the owner of Red Dog Music, a meticulously modelled three-dimensional approximation of an online musical equipment retailer based in Edinburgh.

So what? So stop playing with yourself (fnarr) and get out there and play with others (oo-er), meet like-minded (real, physical) musicians, hang out in your local music shop, go to jam sessions and engage with others, communicate with them and, as NWA would say (or at least sample), *express yourself*. You will learn to communicate more effectively though your music and your music will thus undoubtedly become richer, more interesting and, ultimately, better.

If you would like to air your views in this column, please send your submissions to soundingoff@soundonsound.com or to the postal address listed in the front of the magazine.

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